VŠB – TECHNICAL UNIVERSITY OF OSTRAVA

Faculty of Mechanical Engineering

Department of Applied Mechanics

ANALYSIS OF MINIMUM MEASURABLE NOISE LEVEL USING STANDARD MEASUREMENT EQUIPMENT

Analýza minimální měřitelné hladiny hluku za použití standardních zařízení pro měření

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RANDALL, F. B. Industrial Noise Control and Acoustics. Louisiana: Marcel Dekker, Inc.2003. 534 s.ISBN 0-8247-0701-1
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This Thesis has a context and discussion on the topic of Analysis Of Minimum Measurable Noise level Using Standard measurement Equipment, discussing the issues in measuring the low level sound below the hearing limits in an Anechoic chamber and there is a good precautions taken to avoid all the background noise and then measure the sound with standard measuring microphones and then the noise or traffic is removed using a cross-correlation function, and the results are being discussed with the graphical analysis.

Keywords: self noise, Inherent noise, microphone, Cross-Correlation, Anechoic chamber

ANOTACE DIPLOMOVÉ PRÁCE

BALAJI RAMDAS, Téma diplomové práce: Analýza minimální měřitelné hladiny hluku za použití standardních zařízení pro měření,
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Tato práce má kontext a diskusi na téma Analýza minimální měřitelné hladiny hluku pomocí standardního měřicího zařízení, diskuse o problémech při měření nízkoúrovňového zvuku pod sluchovými limity v Anechoické komoře a jsou přijata dobrá preventivní opatření, aby se zabránilo všem šum pozadí a pak změřit zvuk pomocí standardních měřících mikrofonů a poté se šum nebo provoz odstraní pomocí funkce vzájemné korelace a výsledky jsou diskutovány s grafickou analýzou.

Klíčová slova: vlastní šum, vlastní hluk, mikrofon, křížová korelace, Anechoická komora

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LIST OF SYMBOLS USED

SYMBOL	NAME OF THE QUANTITY (UNITS)
A	Amplitude (m)
t	Time (sec)
ω	Angular frequency (rad/sec)
dB	Decibel
μPa	Micropascal
Hz	Hertz
Pa	Pascals
f	Frequency (Hz)
λ	Wavelength (m)
dB(A)	A Frequency weighting
dB(C)	C Frequency Weighting
dB(Z)	Z Frequency weighting
V	Volts
v	Velocity (m/sec)
Ω	Impedance (N/m/sec)
P	Pressure (Pa)
Po	Standard lowest sound pressure (Pa)

ABBREVATION	FULL NAME
SNR	Signal to noise ration
EIN	Equivalent Input Noise
SPL	Sound Pressure Level
B&K	Brüel&Kjær
PCB	PicoCoulomB
MEMS	Micro electro mechanical systems
GRAS	Generally Recognised as safe

ABSTRACT

Noise is evidence of negative environmental factors which can have a significant impact in measurement sector especially in low pressure level sound measurement. It is a complicated task to measure Stationary Acoustic sound pressure level in any environment with an approach to achieve the desired target which lies below the minimum measurable sound pressure level of the entire measurement system itself, with the influence of self noise or inherent noise of the system, which is generally the noise from the equipments used for the measurement such as microphones and other components used in the measurement system etc. This is performed by certain methodology by using a Standard Measurement Equipment inside an Anechoic chamber and measure the lowest possible measurable sound with assistance of Cross-Correlation method.

1. INTRODUCTION

Measuring extremely low levels of sound pressure is critical for many applications. It can be used in recording studios, auditory spaces, concert halls and sound insulated spaces to identify the acoustic environment. Noise is becoming increasingly relevant in current technological practice and various engineering and research tasks concentrate on minimizing noise levels and eliminating their negative impacts.These may also be used for quiet machines and devices such as lighting-armatures to measure noise emissions. Normally, sound pressure levels are measured using a single measuring microphone connected to a sound level meter or sound analyser. With this setup the lowest sound pressure that can be determined is restricted by the inherent noise of the microphone, preamplifier and instrument.

Most of the high-quality sound level meters and analysers with standard 1⁄2 " measuring microphones have an A-weighted noise floor in the range 14–20 dB re 20 µPa. A 1" microphone device with a noise floor just below 0 dB is available at the cost of reduced environmental stability. This shows, for the time being the drawbacks of conventional measuring technique, but in this experiment we are using the standard measurement equipment not an advanced and hence achieve the desired results. The measurement system's inherent noise contributes to the level to be measured. However if the noise level is known and accounted for, a satisfying test can only be done above the noise floor for levels 2–5 dB, even then we desire to achieve the sound level which has to be measured below 0 dB and to be precise even the lower level of the measuring microphone itself. Hence, we use sound level meter to measure sound pressure levels lower than the underlying noise. This can be achieved by using two channels of measurement and the technique of Cross-correlation. With ordinary measurement microphones, rates below 0 dB can be achieved. As shown in below Figure 1 it explains the hearing range domain with X- axis mentioning the frequency range in (Hz) and Y- axis mentioning Decibels (dB) and the Figure 2 explains the scale of hearing ranges of different fields or domain values mentioning in Decibels (db) on left side and in Pascals (Pa) on the right side, also to mention we are measuring the sound even below the 0 dB scale.

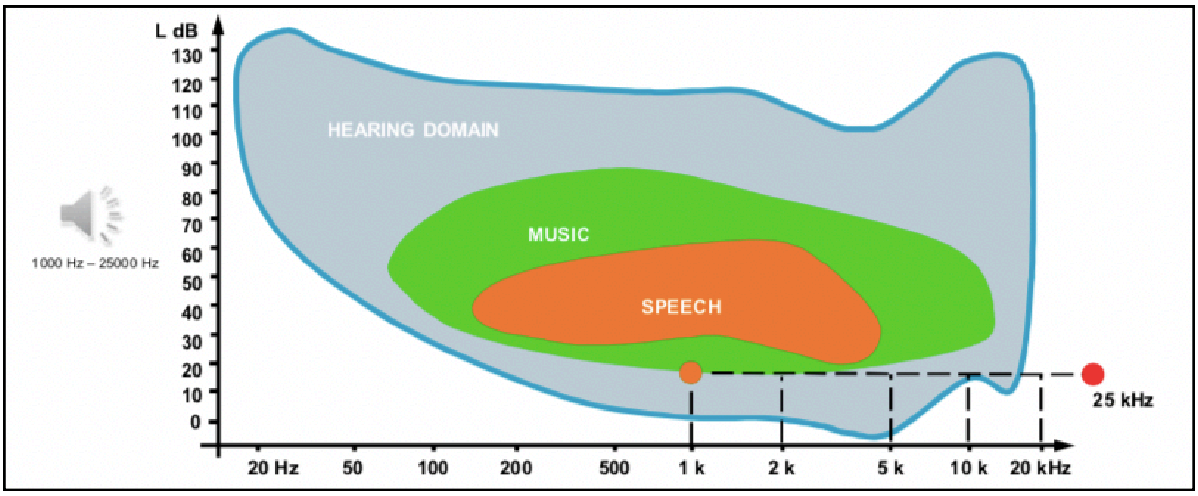

Figure 1 : Domains of the sound levels

The phenomenon of developing consumer product acoustics reflects shifts in legislative conditions that impose increasingly strict noise limits in all areas of human activity. There is an increase in the amount of acoustic testing of series production samples, development in the field of acoustic product properties, along with the development of acoustic properties of products, requiring correct built and properly constructed acoustic measuring equipment. Measures laid down by common standards require acoustic testing in an Anechoic chamber which is especially designed to eliminate the reverberation or in other words lets call it as eliminating echos which can cause an impact in sound measurement and affect the accuracy. Hence, the entire experiment will be carried out in an anechoic chamber. The experiment is carried out with the help of two standard similar microphones and and analyser and computer to analyse the results and we use a method to compute the results that is Cross-Correlation. Rate measurements Based on the technique of correlation The inherent noise affecting the lower end of the measuring range is usually the random noise in the electronic circuitry and the resistive portion of the microphone impedance. This means that measurements would be constrained by an equal noise cycle if the same acoustic field is measured with two similar measuring systems. The signals from the two noise processes should be separate and uncorrelated, even though they are of equal duration. The signal portion that originates from the acoustic field will, however, be identical and hence completely correlated. Therefore, the acoustic signal and the noise can be distinguished by the Cross-correlation technique

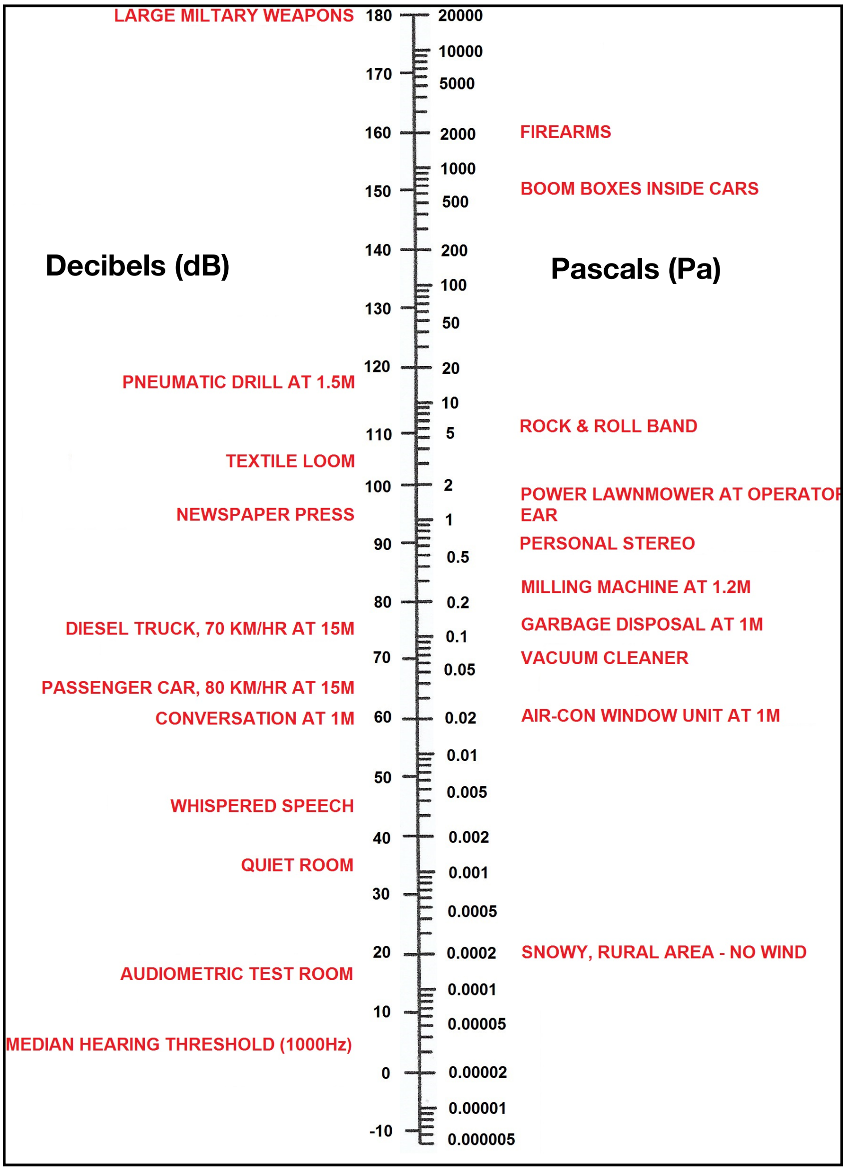

Figure 2 : Scale of sounds level from different sectors dB(RHS), Pa(LHS)

2. PHYSICS OF SOUND

Noise can be characterized as "unpleasant or unwanted sound," or any other disturbance. From the point of view of acoustics, sound and noise signify the same phenomenon of changes in air pressure over mean atmospheric pressure; the distinction is highly subjective. To anyone else, what is sound may very well be noise. Recognition of noise as a significant threat to health is a modern-day development. Modern technology has intensified the multitude of causes of noise-induced hearing loss; loud music also takes its toll. Though amplified music can be regarded as sound (not noise) and to attract many but most of industry's needless noise actually brings very little satisfaction, or none at all.

Sound (or noise) is the product of fluctuations in pressure or oscillations in an elastic medium (e.g., air, water, solids), induced by a vibrating surface or turbulent flow of fluids. Sound propagates in the form of longitudinal waves (as opposed to transverse waves), generating a series of elastic medium compressions and rarefactions. As a sound wave propagates through air the frequency oscillations are above and below the ambient pressure in the area [3].

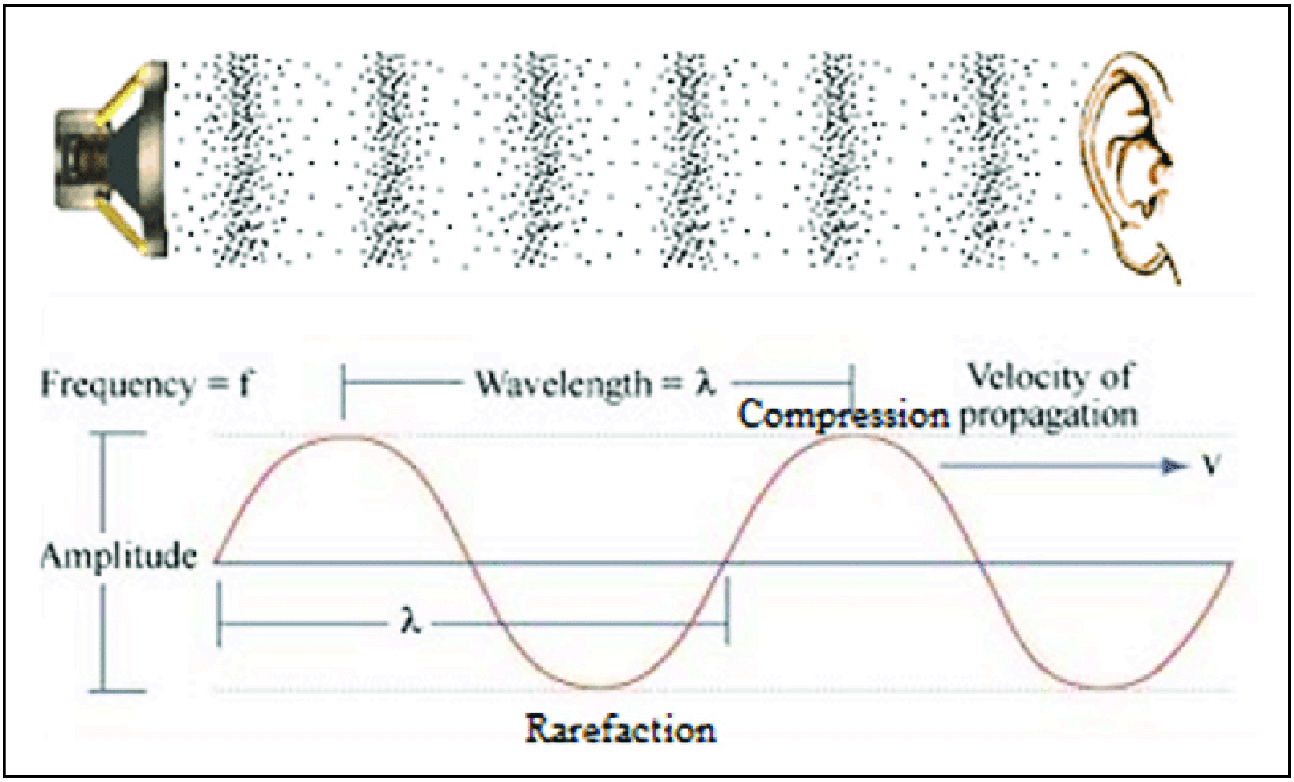

Figure 3 : propagation of sound

3. FUNDAMENTALS OF SOUND

The explanation of the terminologies which define the sound and its physics are discussed as follows [4],

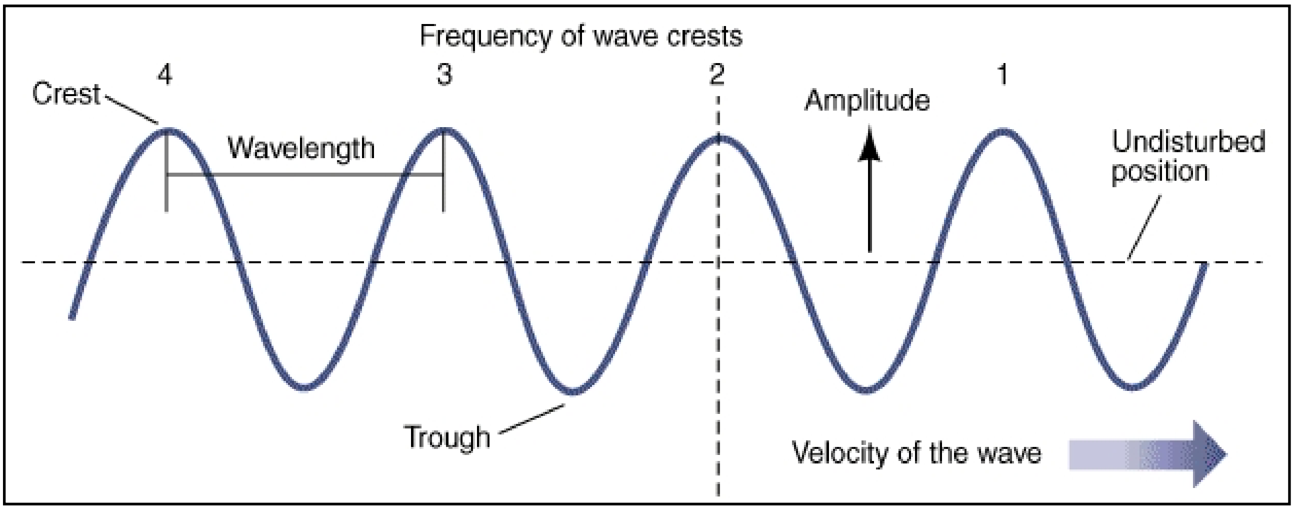

Figure 4: Sin wave with the defining parameters

3.1 Amplitude is a wave's fluctuation or distance from its mean value. It's the degree to which air particles are replaced by sound waves, and this amplitude of sound or sound amplitude is known as sound loudness.

3.2 Wavelength (λ)is the difference between the equivalent opposite parts of a wave. The frequency is the same as the source frequency, which is the number of waves passing one point per unit.

3.3 Frequency (f) is the number of cycles of change of pressure in the medium per unit time, or simply the number of cycles per second, expressed in Hertz (Hz)

3.4 Period (T) which is the time taken to reach a fixed point for one wave cycle.It is related to frequency by: $T = 1/f$

3.5 Sound is an audible, mechanical vibration, moving through any physical medium as pressure oscillates; solid, liquid, or gas. In psychology and physiology, sound is the brain's reception of such pressure oscillations, and such perception.

3.6 Noise is unwanted sound considered irritating, distracting or harmful to hearing. From the point of view of physics, noise is indistinguishable from sound, as both are vibrations through a medium, such as air or water. The difference occurs as the brain hears a sound and perceives it.

3.7 Sound speed is the speed at which a sound wave's compressions or rarefactions move in the sound propagation direction. The speed of sound should not be confused with the velocity of particles. The sound wave travels fairly quickly, while the particles oscillate at a relatively low particle velocity about their original location. At normal ambient conditions (20 ° C and 101.325kPa) the speed of sound in air is 343 m/s.

3.8 Acoustics is sound science. This specialty comprises infrasonic, ultrasonic, and audible frequency ranges of sounds and vibrations. Acoustics subcategories include: Aero-acoustics, bio-acoustics, psychoacoustics, music theory, noise reduction, speech pathology, undersea acoustics, and vibration. A multitude of tests and analyses are conducted within each of those subcategories.

3.9 Sound pressure is the local pressure difference caused by a sound wave from the ambient atmospheric journal. The Pascal (Pa) is the unit SI (metric) for sound intensity. Under air a microphone is used to measure sound intensity. This is weighed in water with a hydrophone. Sound is usually processed by the ear by humans and other creatures.

4. SOUND FIELDS DEFINITION

The technical name given to the dispersion of sound energy within defined boundaries is a sound field.

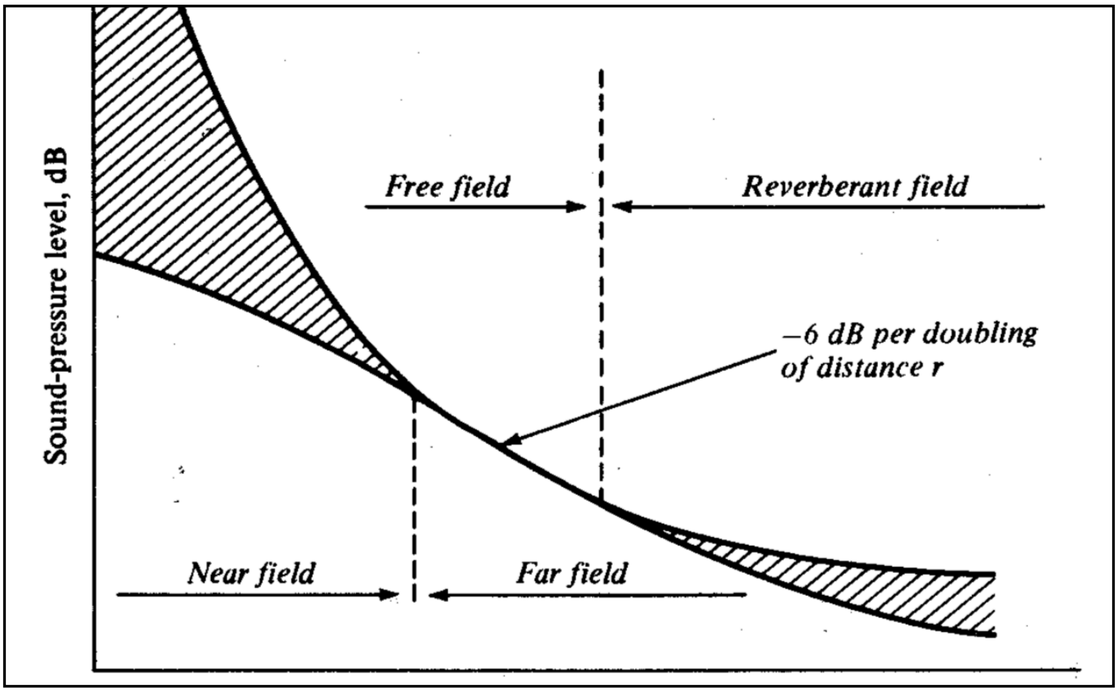

Figure 5: Explanation of Different sound Fields

4.1 FREE FIELD

The free field is a region in space where sound can spread free from interference of any kind. A free field is a homogeneous, boundary-free medium or reflecting surfaces. In view of the simplest type of a sound source that would radiate sound equally from an apparent point in all directions, the energy emitted at a given time would propagate in all direction[4], [5].

4.2 NEAR FIELD

A source's near field is the area surrounding a source in which the sound intensity and the velocity of the acoustic particles are not in motion. In this region the sound field does not decrease by 6 dB each time the distance from the source is increased. The near field is limited to a distance of approximately one wavelength of sound from the source, or equal to three times the largest dimension of the sound source.

4.3 FAR FIELD

A source's far field starts where the near field ends and stretches to infinity. Remember that in the transition area, the transition from near far field is gradual. In the far field, each time the distance from the source is multiplied, the direct field radiated by most computer sources will decay at a rate of 6 dB. The decay rate for line sources like road noise ranges from 3 to 4 dB.

4.4 DIRECT FIELD

A sound source's direct field is defined as that part of the sound field which has not suffered any reflection from any surfaces or obstacles in the space.

4.5 REVERBERANT FIELD

This field this the most important on our discussion as we are more concerned about this field regarding this experiment is all about. Once sound waves encounter an obstacle, such as putting a source of noise within limits, part of the acoustic energy is reflected, part is absorbed and part is transmitted. The relative quantities of acoustic energy that have been reflected, absorbed, and transmitted depend significantly on the existence of the barrier. Different surfaces have different ways to reflect, absorb, and relay the sound wave of an event. A smooth, compact, flat surface. The acoustic energy can reflect much more and absorb much less than the porous and soft surface [3].

When a room's boundary surfaces consist of a material that reflects the sound of the event, the sound created by a source within the room, the direct sound may bounce from one boundary to another, giving origin to the sound reflected. The higher the proportion of the sound reflected by the accident, the greater the ratio of the sound reflected to the overall sound in the closed room. And after the noise source has been switched off this "built-up" noise will continue. This phenomenon is called reverberation and the environment in which it occurs is called a reverberating sound field in which the noise level is not only dependent on the acoustic power radiated. Yet also on space size and the acoustic boundary absorption properties.As the surfaces become less reflective and more noise absorbent, the reflected noise becomes less, and the situation tends to a condition of "free field" where the only significant sound is the direct sound. It is possible to arrive at sound propagation characteristics similar to free field conditions by covering the boundaries of a limited space with materials which have a very high absorption coefficient. Such a space is called an anechoic chamber, and these chambers are used for acoustic research and measurement of sound power. In action, at any reflection there is still some absorption and so most work spaces can be regarded as semi-reverberant.

The reverberation phenomenon has little influence in the region very close to the source, where direct sound is dominant. However, far from the source, and unless the walls are rather absorbent, the reflected, or indirect, sound can significantly affect the noise level. The sound pressure level in a room can be perceived as a combination of the direct field (sound radiated directly from the source before a reflection) and the reverberant field (sound reflected at least once from a surface).

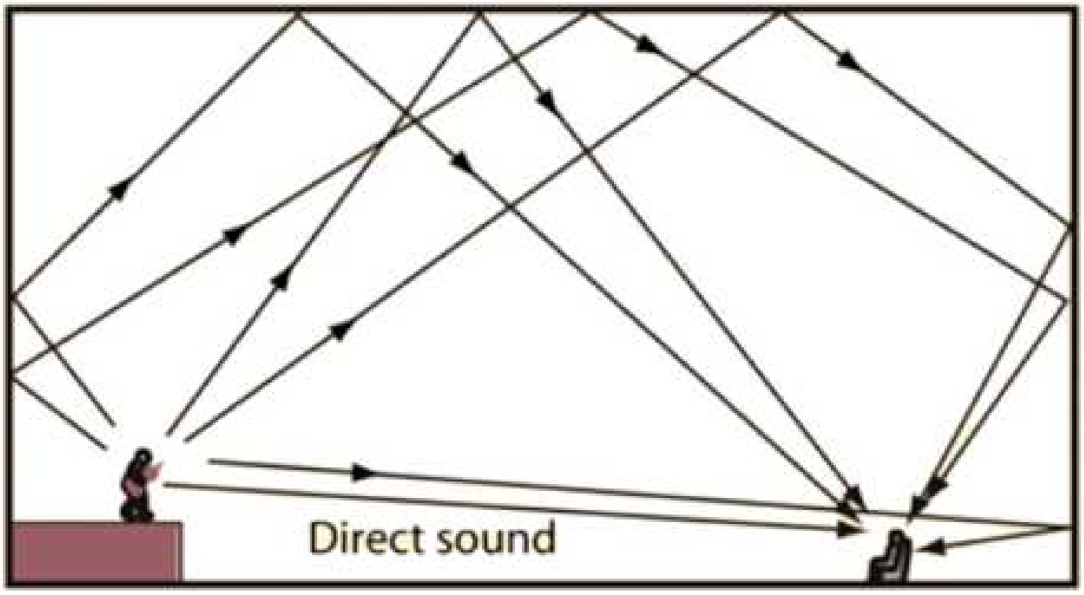

Figure 6 : Reverberant field

5. ISSUES IN SOUND MEASUREMENT

The explanation of the terms which define the sound and its physics are discussed. Sound measurements are always made with measurements of noise. All transducer types and data acquisition circuits introduce different noise types which make the sound measurement less accurate. The environmental noise often disturbs the precision of the study by sound detection, but often the adjustment in ambient noise contributes to useful details. Functional Measuring Devices In practical measurements the detection of a sound caused by a fault will be difficult because of the background noise. The noise floor is the level of background noise in a signal. Just signals above the noise floor can be detected, they contain continuous noise but random noise also. Because noises cannot be fore seen. The measurement condition must be the same, temperature, load, background noise. If a fault occurs, the vibration, the noise produced overlap with the base sound signal, and a modulation appears. In case of different faults frequency and amplitude modulations appear. The last can be easy detected due to the sideband frequencies in the frequency spectrum and cross correlation

By considering all these factors all these factors there should be a very precautionary measures in measuring the sound how ever there are some un-controllable factors which influence the sound measurement which is Inherent noise.

5.1 SELF-NOISE / INHERENT NOISE : Inherent noise of the measurement system as the unavoidable uncertainties even with ideal measurement where no systematic error exists. Inherent noise is basically unavoidable since it originates from the nature of molecular-level interactions. On the basis of measurement the inherent noise should be as low as possible

Every microphone produce a certain noise level through its electronics, its transducer part, and it's housing. The intrinsic noise is called Inherent noise or self-noise. It is a familiar sound for anybody who uses a cell phone, for example, leads to the hiss which is audible when your cell phone is on and no one is speaking. Self-noise is an ever-present limitation . The aim is to make the microphone transmit as much signal to the rest of the signal chain as much as possible. But part of the signal that you catch from the audio source will come under the intrinsic noise of the microphone, which is often called the floor noise level. The noisier a microphone is, the less of a signal you've got. A microphone with lower noise will allow you space to separate the sound you want from the noise you don't. As a consequence, when you start with a quieter microphone the output from the signal chain sounds much better[7].

A high signal-to-noise ratio (SNR) implies a quiet microphone, and a lower SNR specification informs you that the microphone has more self-noise. The self noise, or noise floor, of the microphone does a lot to define the quality of audio you are able to capture and pass onto the signal chain. Signal-to-noise ratio (SNR) and equivalent input noise (EIN) are two specifications that describe where that noise floor is.

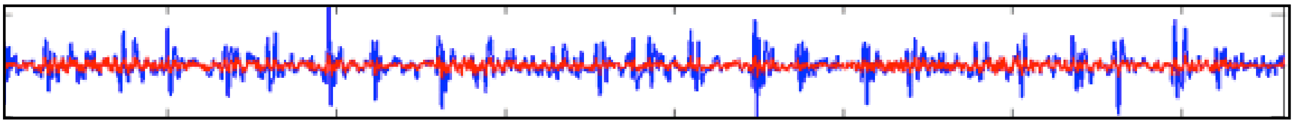

Figure 7 : Sound signal with inherent noise

5.1.1 SIGNAL TO NOISE RATIO (SNR) : Signal-to-noise ratio is a measurement technique used in engineering that compares a target signal to background noise level. SNR is known as the signal-to-noise ratio, often expressed in decibels. A ratio greater than 1:1 corresponds to more signal than noise.When comparing the different microphones SNR, we should make sure they are based on the same weighting and bandwidth. A comparison is not valid when the calculations use the same weighting and bandwidth[7].
Signal-to-noise ratio is defined as the ratio of the power of a signal to the power of noise :

$$\mathrm{SNR} = \frac{P_{\mathrm{signal}}}{P_{\mathrm{noise}}}$$

5.1.2 EQUIVALENT INPUT NOISE (EIN) : Equivalent input, as if it were an input to the same device, is a way of referring to the signal or noise level at the source. This is done by eliminating all changes in signal to get the units matching the data.While eventually you'll find more than one dimension of the output of a microphone, a low noise floor is a make or-break spec for difficult audio capture applications. And if you used to need high SNR, there should be a defining and controlled environment as well as the precise equipments should be chosen for the test to be carried out[7].

5.2 EFFECT OF BACKGROUND NOISE

Ideally, when a source of noise is measured, the measurement will only evaluate the actual air-borne sound from the source, without any appreciable input from other sources of noise. The measuring room can need to be shielded from external noise and vibration to ensure separation from other sources. If, relative to the ambient sound pressure level alone, the change in sound pressure level in any given unit, with the sound source working, the sound pressure level due to both the sound source and ambient sound is exactly the sound pressure level due to the sound source[9].

When evaluating equipment noise, the background noise level in each band should also be evaluated to determine if the difference in the band levels is greater for the overall noise and background. The background noise range is usually different from that of the noise to be measured. If this difference between total noise level and background level is less, an attempt to lower the background level should be made. Usually the first step is to work directly on the source or sources of this background noise. The second step is to focus on the route of transmission between the source and the measuring point. This move can simply mean closing the doors and windows, whether the source of intrusion is outside the building, or barriers are erected, apply acoustic treatment to the room and, if the source is in the house, open the doors and windows. The third step is for the measurement method to increase the difference. A point closer to the instrument may be chosen, or an analysis of the background noise field can indicate that the place of measurement may be moved to a minimum of that noise. The latter scenario is more probable when an experiment is done, and the noise level is exceptionally high in a specific band [4].

6. ANECHOIC CHAMBER

On the analysis of the issues in low level sound measurement, an Anechoic chamber is being opted for the measurement os the sound and carry out the experiment as to be discussed in this project.

One of the key problems involved in characterizing anechoic chambers is determining the same coefficient of absorption or reverberation time. The conventional approaches only aim to check chamber well-being to the degree that the inverse squared law of this sort of enclosure agrees. Such approaches are based only on calculating the amount of sound pressure within the enclosure. The idea that becomes in this work aims to obtain data on the absorption of anechoic chambers through the transfer functions of microphone pairs, or through the study of the impulse response between pairs of microphones. Based on the results of the transfer functions between pairs of microphones, the inverse squared law can be easily tested to allow the chamber cut-off frequency to be calculated. The anechoic chambers qualification could be verified by doing a band filtering [1].

An acoustic anechoic chamber is a shielded room constructed under conditions close to free space for performing sound measurements. Issues relating to insulation, absorption, and construction are discussed and answered, including preliminary findings and recommendations for the purpose of the experiment. Acoustic anechoic chambers are environments with a high acoustic isolation from the surrounding environment, used for measuring systems under conditions close to free space. Anechoic chambers need to be of substantial size due to the basic specifications of these measurements, in order to produce a reasonable response. This contributes to high construction and maintenance costs, and the need of an adequate physical space.Moreover, if an anechoic chamber is designed for a particular set of applications, it can overcome many of those limitations. In the context of our group's hearing aid project, a chamber was needed to calculate the response of two microphones from a directional source, likely not matched [1],[2].

Figure 8 : Anechoic chamber in VŠB-TUO

6.1 CHAMBER DESIGN PARAMETERS

Anechoic chamber is designed in such a way that it has to satisfy nor justify some set of parameters to ensure the precision of the results obtained and they are discussed as follows[6].

6.1.1 SHAPE

Large anechoic chambers are typically designed in the form of a rectangular cuboid, contributing mainly to architectural constraints. While this configuration is sensitive to standing waves, the basic resonance frequency is generally small enough to be ignored.

This effect is not negligible in the case of a smaller chamber so a non-regular shape is preferred. Despite this recommendation a rectangular cuboid has been chosen as the design shape, as it simplifies simulation and design.

6.1.2 DIMENSION

Chamber sizes are a key factor in architecture. The standing wave problem can be mitigated if wisely selected. A common scale was multiplied by three small prime numbers for determining the internal width, height, and length. This achieves a combination of standing wave modes which do not overlap in interest frequency range. Anechoic room was designed according to the CSN EN ISO 3745 series of standards.

6.1.3 INSULATION

To determine the appropriate acoustic insulation, Sound level meter Brüel&Kjær 2250 was used to determine the maximum SPL level at the site of our sound lab. The average SPL was determined to be 60 dB(A) in interest frequency band

Originally tests were performed with a single wall insulation. If wave fronts are believed to be smooth, interfaces infinite and no dissipation occurs, the loss of transmission at the interfaces can be determined by

$$TL = -10\ log10\ (| T |^2)$$

Where, (T) is the quotient of two phasers which reflect the complex amplitude of the incident and the pressure waves transmitted.

The resulting width was unacceptably wide, so it was called a double wall insulation. In this setup, simulations were repeated, and the wall widths and spacing differed. The best values for these parameters were determined by limiting the first high-frequency transmission zero above the interest frequency of the maxi-mum and forcing the low-frequency zero to a minimum value. The values found for wedges are one meter long with an air gap of 10 cm in air gap.

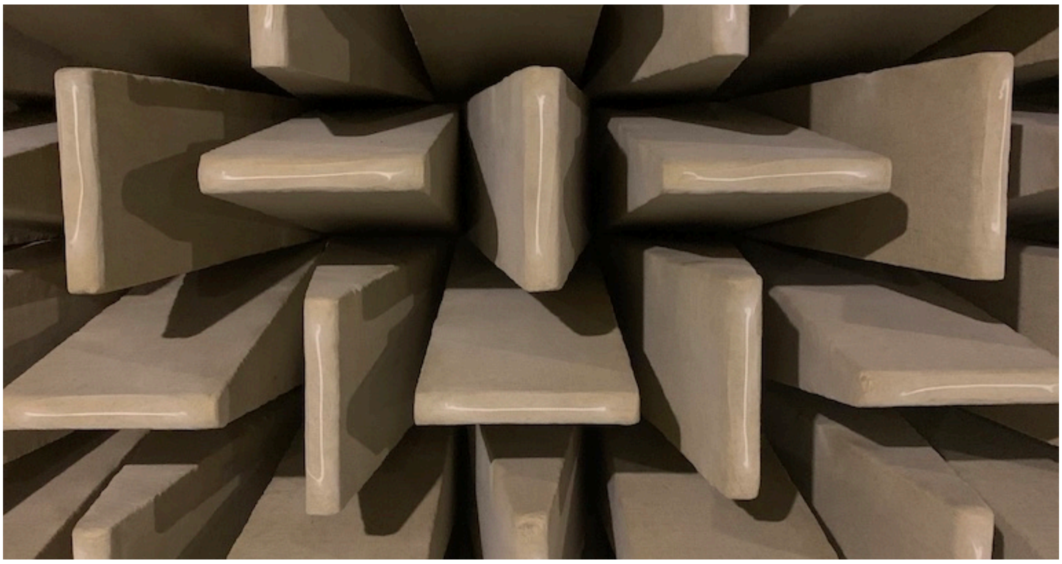

Figure 9 : Alignment of Wedges in the anechoic chamber with air gap

6.1.4 ABSORPTION

An internal wall lining is necessary for minimizing echoes inside the chamber. Initially a survey was conducted on anechoic wedges. Several factors were considered: the frequency of the absorption cut-off, the coefficients of absorption, content, dimensions, shape and protection. None of the products available on the local market were satisfactory, and it was determined prohibitive to import for Eigen content. So it was decided to design and create our own anechoic wedges.

Figure 10: Wedge from Anechoic Chamber

Acoustic pyramids for eliminating sound echoes at chamber boundaries. While for the traditional chamber the acoustic wedge is prevalent in shape, the pyramid figure is used due to the volume-wise cost-efficiency. The pyramid occupies 33 percent less volume than the wedge body with

Figure 11: Pyramid Structure Wedge

similar structural parameters, therefore with some degree of output loss the construction cost to the chamber.

Room Dimensions [Width; Length; Height]	9,5 m; 8,5 m; 2,6 m
Frequency Attenuation	from 100 Hz
Absorbent Material	Porous mineral fiber material

Table 1: parameters of Anechoic chamber in VŠB-TUO

7. SELECTION OF MICROPHONES

Selection of microphone for this experiment is the most important vital decision, not every microphone can be used to ease the low level sound satisfying all the basic protocols and most importantly the whole experiment is result oriented, and the obtained results to be as precise as possible and the main aim is to achieve accuracy in low level sound measurement.

The output of a microphone is defined in several specifications. This section describes the most common parameters used to assess if a microphone is appropriate for use in a given application.Measurement microphones are devices that are used to measure sound pressure in air. The sound pressure is used to determine a lot of sound field characteristics. Basic measures of sound pressure are used to learn about sound sources and the effects of the sound on sound field structures. Microphones for test and measurement applications have a flat frequency response, good sensitivity, robustness to the environment and high stability. Measurement microphones do have a well-defined method of calibration which is traceable to the basic quantities of physics.

A microphone's sensitivity is the ratio of the output voltage to the sound pressure at the origin. This is usually reported in precision measuring microphones at a common reference frequency of 250 Hz. Sensitivity is expressed in volts per Pascal (mV / Pa) while in decibels a sensitivity point corresponds to one volt per Pascal (dB re 1V / Pa). When no external stimuli are present, self-noise or endogenous noise is detectable by a microphone or microphone device. The output is due to the diaphragm thermal motion in a microphone cartridge, and the microphone device preamplifier noise. The combined device is related to a significant limiting factor in microphone noise which is being discussed in the issues already.

Microphones are selected based on the quality of its inherent noises, as lower the inherent noise better for the measurement, and Inherent noise of the measurement system as the unavoidable uncertainties even with ideal measurement where no systematic error exists. Inherent noise is basically unavoidable since it originates from the nature of molecular-level interactions. On the basis of measurement the inherent noise should be as low as possible [11].

Selection of Microphones is based on the following factors,

- Indoors or outdoors
- Duration of test

- Background noise, and frequencies of background noise
- Humidity of the test area
- Temperature in test area
- Location of the test equipments
- Positioning of the test equipments
- Characteristics of surrounding and equipments
- Proximity of surrounding test equipments
- Minimum and maximum frequency and amplitudes required

7.1 TYPESOF MICROPHONES

There are several types of microphones available on their discussion we can conclude what type of microphone is suitable for the experiment to be carried out and they are discussed as follows

7.1.1 Precision condenser microphones

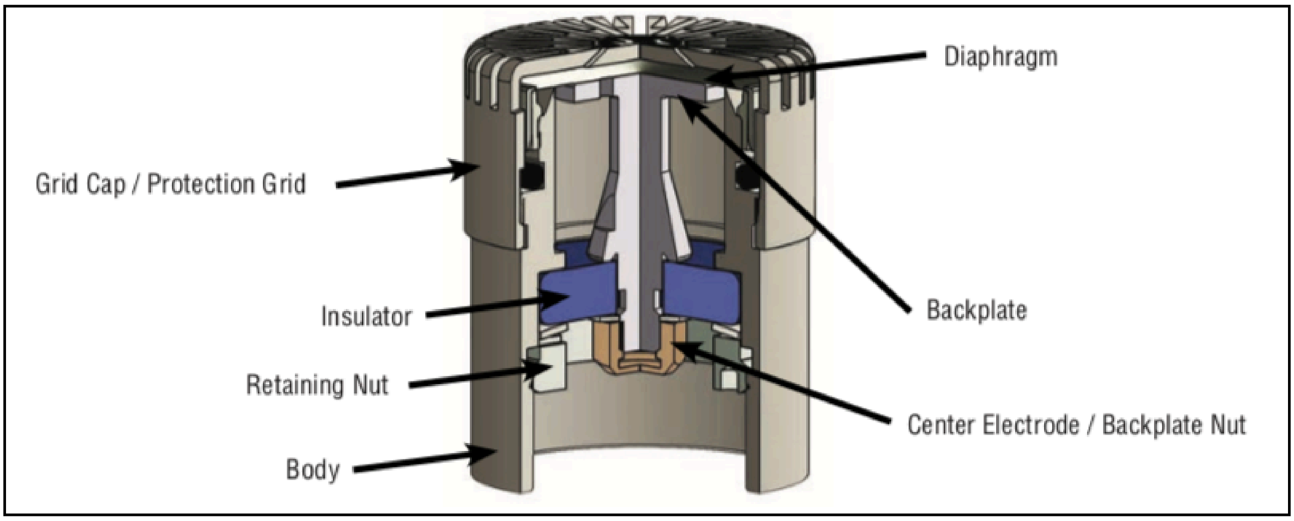

Figure 12: Cross section of standard Condenser microphone

This type of microphones consist of a thin metal membrane in close proximity to a solid metal plate. This forms a variable condenser that converts diaphragm motion to voltage. When the microphone is subjected to pressure changes, the resulting diaphragm motion induces a change in the microphone's capacitance. The displacement of the diaphragm is directly proportional to the pressure from the exposed tone. The best materials for effective acoustic research are the precision condenser microphones. They're very stable with variations in time and temperature or humidity. To suit a wide variety of preamplifiers and portable calibrators, the cartridges are standardized. They're the most widely used microphone type for precision sound level meters.

A constant charge is applied on the backplate to measure the varying capacitance of the microphone due to the changing sound intensity. A voltage is used in some condenser microphones to create a difference in charge between the backplate and the diaphragm. There are microphones which are externally polarised. The diaphragm motion (specifically the minute change in the distance due to sound pressure) is proportional to the change in voltage across the terminals of the microphones. The microphone's output voltage is directly proportional to the sound pressure on the diaphragm.

7.1.2 Array microphones

Figure 13: Array Microphone from PCB

This type of microphone use either a cartridge in the electret style or a MEMS microphone cartridge similar to that used in small electronics. The benefit of array type microphones is that at reduced cost, they are easily mass-produced. While an array microphone's dynamic range and frequency response isn't as flat as a condenser microphone, the low cost makes it an excellent option for wide channel count testing. For ease of use, preamplifiers are permanently matted to the collection microphones.

7.1.3 Piezoelectric microphones

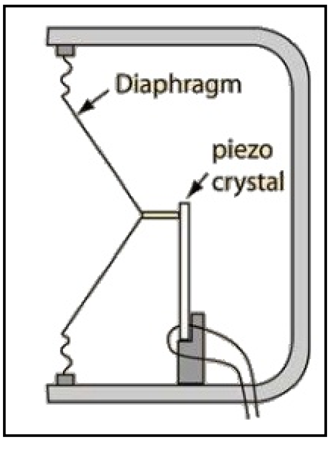

Figure 14 : constructional representation of Piezoelectric Microphone

Quartz or ceramic crystal is used as a sensory feature by Piezoelectric or dynamic pressure transducer. The crystal charge production is proportional to the position of stress on the crystal. These dynamic pressure transducers that have an integrated amplifier with an ICP design, allowing the sensor to output a voltage that is proportional to the pressure input. Though these acoustic sensors have very low levels of sensitivity, they are very robust and measure very high pressure of amplitude (170 dB or higher). As a result, on piezoelectric microphones the noise floor level is very high (greater than 70 dB) vs condenser and microphone collection. This concept can be used for shock, blasting and very high acoustic pressure measurement techniques.

7.1.4 Surface microphones

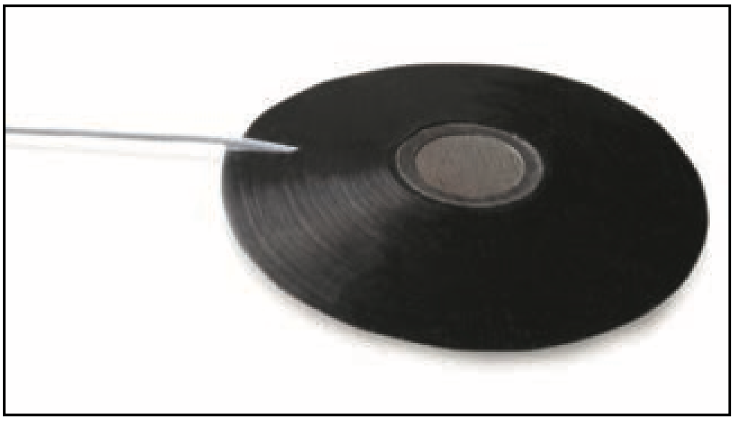

Figure 15: Surface Microphone from PCB

This type of microphones are used for surface pressure and noise measurements in confined areas. To minimize wind-induced noise a surface microphone should have a faring built and optimized. The versatile architecture of the faring allows to be placed flush or adhesively on flat or curved surfaces. A surface microphone's low profile allows for noise measurements which minimize interference with the sound field.

7.1.5 Probe microphones

Figure 16: Probe Microphone from GRAS

They are designed for acoustic measurement in: low, hard-to-reach places, close to field acoustic testing, narrow areas and extremely high temperature environments (up to 800 ° C). Measurements are achieved by insulating the sensor with a thin tube from the source. In most field types sample microphones are tuned to have a flat frequency response. The small size of the probe makes it less obstructive irrespective of what field of sound it is mounted.

7.1.6 Externally Polarized microphones

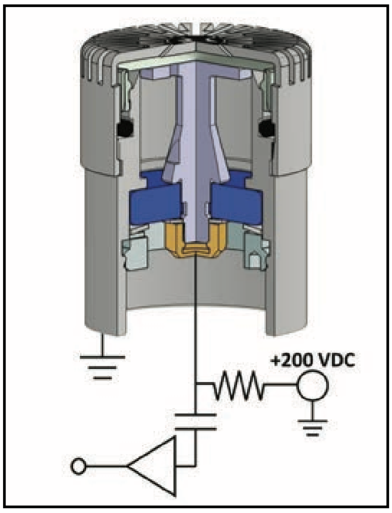

Figure 17: Externally polarized Microphone

Figure 17 illustrates that these microphones have 200 volts positively added directly to the backplate. This needs additional preamplifier insulation to prevent overloading of the device. External polarisation makes it difficult to operate modular devices, such as sound level meters. To control the preamplifier and the separate polarization voltage needed to operate the microphone they require advanced signal conditioning and cabling.

7.1.7 Pre-Polarized microphones

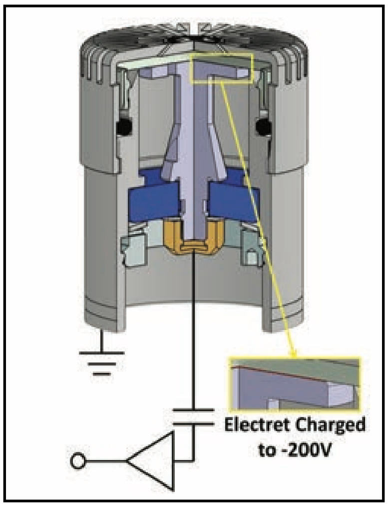

Figure 18: Pre-Polarized Microphone

This Type of Microphone Using a permanent charge equal to 200 volts on the backplate of the microphone embedded in an electret material. It simplifies the preamplifier configuration by eliminating the polarization voltage cabling and making two wire microphone systems possible. It allows lower complexity in the circuitry of the sound level meters. Prepolarized systems are easily interchanged with other sensors for testing and measuring and can be used with data acquisition systems which also support externally polarized microphones. Due mainly to their low cost per channel and interchangeability with other test and measurement sensor cables and power supplies, prepolarized microphones are becoming increasingly common.

Once exposed to the same auditory stimulus they will have opposite stages, a big difference between externally and prepolarized microphones happens. A prepolarized microphone has an equivalent voltage of -200 volts, and external polarization supplies + 200 volts. For fact, when making a test with an externally polarized microphone, the output voltage step is opposite to the sound pressure step. With externally polarized microphones, positive pressure on the diaphragm produces a negative output voltage and a positive output voltage with prepolarized microphones.

7.1.8 Low noise microphones

Are condenser microphones designed to have the characteristic of very low inherent noise. Such noise reduction happens at device level for the microphone. The cartridge has a frequency response which reduces inherent noise from thermal sources. The preamplifier also increases the total device gain and boosts the signal-to-noise ratio for every data acquisition device.

Finally, a microphone with the lowest available inherent noise specification will be chosen. Ideally, Based on the research according to the goal that has to be achieved or precisely according to the accuracy which has to be achieved

Hence, based on the goal and aim of this project is to achieve the lowest sound measured using the standard measurement equipment and not using the highly equipped and advanced Microphone to achieve the results, so the selected Microphone is Brüel&Kjær 1⁄2″ Freefield Prepolarized Microphone,Type 4189. At higher frequencies, reflections and diffractions in front of a microphone's diaphragm cause pressure to increase. For this will result in an increase in output voltage if not corrected. A free-field optimisation means that the microphone's frequency response has been configured to make the free-field response flat at 0 degrees incidence. This microphone is designed for use with the in place security panel. Type 4189 is ideal for use in class 1 Sound Level Meters and for all high-precision acoustic measurements requiring a powerful and secure free-field microphone with a 20 kHz upper frequency [11].

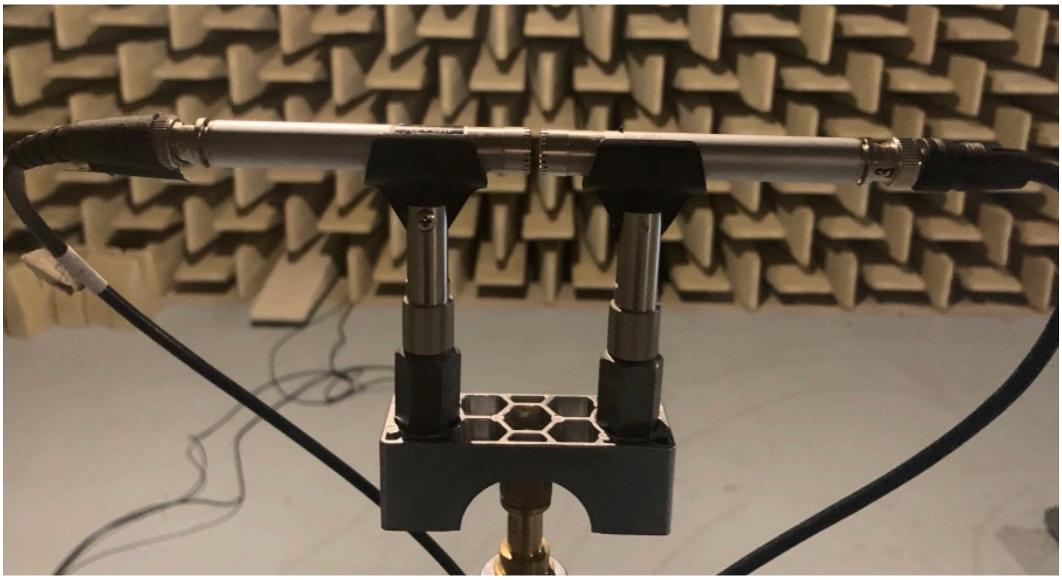

Figure 19: Selected Microphone set up

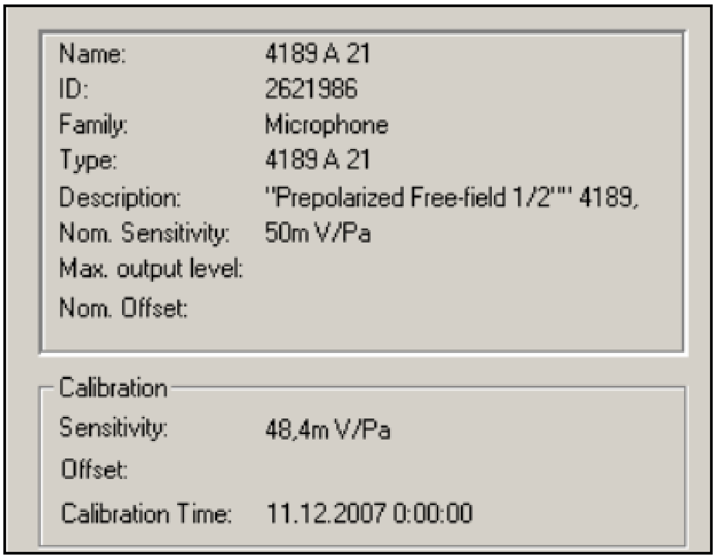

Figure 20: Specification of the selected B&K Microphone

7.2 PRE-AMPLIFIER INTRODUCTION

A preamplifier's primary function is to remove the signal from the detector without reducing the intrinsic signal-to-noise ratio significantly. The preamplifier is therefore placed as similar to the detector as possible and the input circuits are built to suit the detector's characteristics. Based on whether the air is used, various pulse processing methods are usually used. There are three type of Preamplifiers namely,

7.2.1 CURRENT-SENSITIVE PREAMPLIFIER

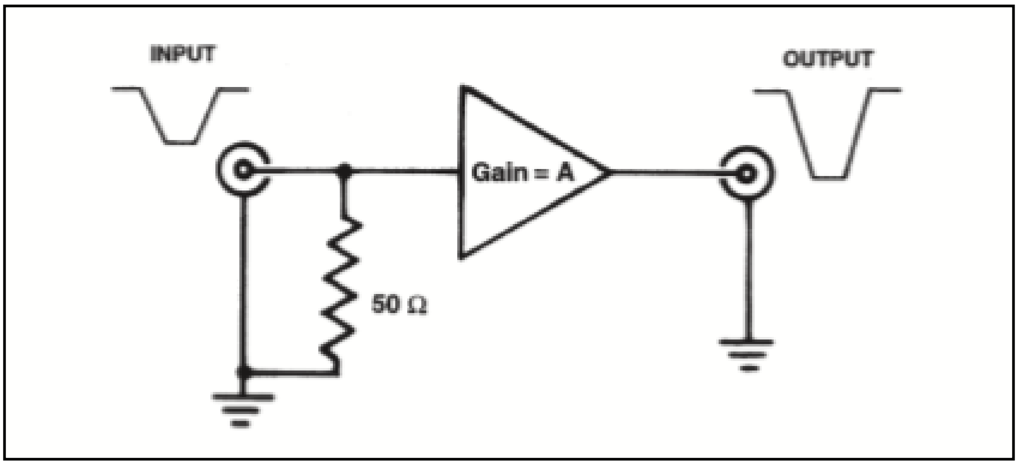

Figure 21: Current-Sensitive Preamplifier

Several detector types, such as photomultiplier tubes and micro-channel plates, generate a moderately large and fast-rising output signal through a high output impedance. Pulse processing for timing or counting with these detectors can be rather simple. A properly- terminated 50-Ω coaxial cable is attached to the detector output, so that the current pulse from the detector develops the desired voltage pulse across the 50-Ω load presented by the cable. For scintillators mounted on 14-stage photomultiplier tubes, this voltage signal is usually large enough to drive the input of a fast discriminator without further amplification. For single-photon counting, 10- stage photomultiplier tubes, or micro-channel plate PMTs, additional amplification is needed between the detector and the discriminator, and this is the function of the current-sensitive preamplifier[7].

7.2.2 PARASITIC-CAPACITANCE PREAMPLIFIERS

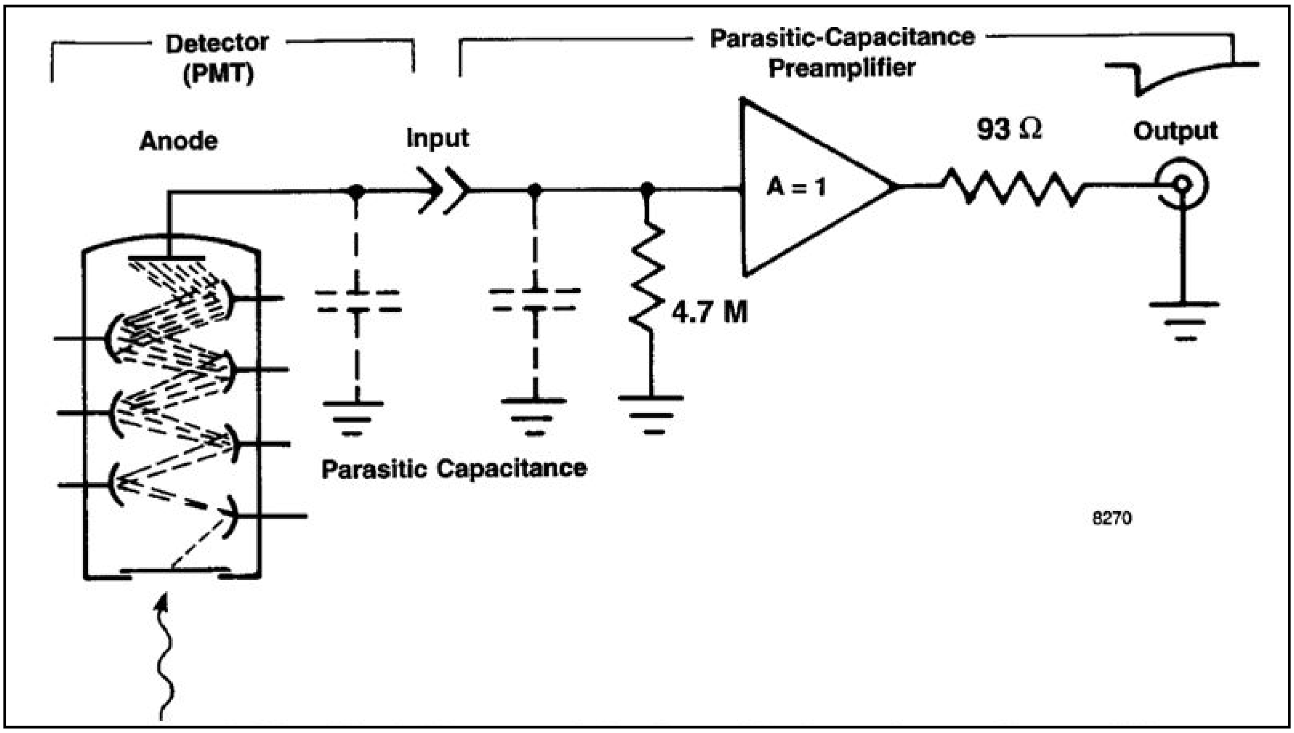

Figure 22: Parasitic- Capacitance Preamplifier

Parasitic-capacitance preamplifiers have a high input impedance(~5 MΩ). Hence, the current pulse generated by the detector is integrated on the combined parasitic capacitance present at the detector output and the preamplifier input. This combined capacitance is typically 10 to 50 pF. The resulting signal is a voltage pulse having an amplitude proportional to the total charge in the detector pulse, and a rise time equal to the duration of the detector current pulse. A resistor connected in parallel with the input capacitance causes an exponential decay of the pulse with a time constant ~50 µs. An amplifier having a high input impedance and unity gain is included as a buffer to drive the low impedance of a coaxial cable at the output. The 93-Ω resistor in series with the output absorbs reflected pulses in long cables by terminating the cable in its characteristic impedance. Parasitic-capacitance preamplifiers are not used with semi-conductor detectors because the gain of this type of preamplifier is sensitive to small changes in the parasitic capacitance. For partially-depleted semiconductor detectors the detector capacitance varies with the bias voltage applied to the detector diode

7.2.3 CHARGE-SENSITIVE PREAMPLIFIER

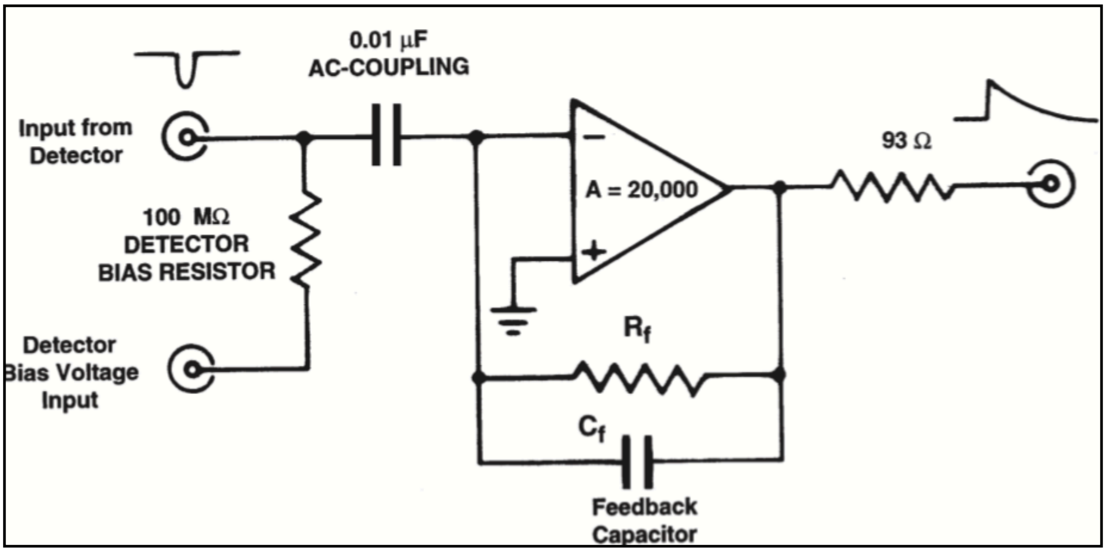

Figure 23: Charge-Sensitive Preamplifier

These preamplifiers are preferred for most energy spectroscopy applications. The signal from a semiconductor detector or ion chamber is a quantity of charge delivered as a current pulse lasting from 10^{-9} to 10^{-5} s, depending on the type of detector and its size.For most applications the parameters of interest are the quantity of charge and/or the time of occurrence of an event. A charge-sensitive preamplifier can deliver either or both. Because it integrates the charge on the feedback capacitor, its gain is not sensitive to a change in detector capacitance, and in the ideal case, the rise time of the output pulse is equal to the detector current pulse width.

8. ANALYTICAL SOLUTION

Even after taking the precautions to avoid the noise during the experiment to achieve accuracy, There is self- noise or Inherent noise which is being added to the measured sound signal. We use a method called Cross-Correlation to eliminate the noise which is being added to the measured sounds signal by two microphones and we eliminate the noise from both the microphone's self noise to obtain the accurate sound signal.

8.1 CROSS-CORRELATION

Cross-correlation (or cross-covariance) consists between two signals on the displaced dot product. The degree of similarity or interdependence between two signals is often quantified. In our case, because all measurements were reported using digital acquisition systems, all signals under analysis were evaluated in a discrete amount of time, such that the correlation between two signals x and y of the same N sample length is represented by the following expression:

$$Corr\{x, y\}[n] = \sum_{m=1}^{N} x[m] \cdot y[m+n]$$

If we do y = x we obtain the signal x cross-correlation,

Shows the signals used in those studies: clicks, sweeps. On top of that, there are these ideal signals in the time domain, that is, the electric signal generator equipment producing the signals. The frequency of each signal can be seen in the middle section, where you can understand the different bandwidths. At the edge, each signal's autocorrelations indicate that the higher bandwidth signals have a peak of narrower correlation, so they're easier to detect in general. To understand the importance and usability of each detector using these signals. The results of applying this equation to the results of the obtained correlations are described in the following pages, compared with values obtained by applying time and frequency domain methods. Additionally, the gains achieved by using this technique are also seen in terms of accuracy of detection in various acoustic settings.

8.2 PROCESS OF REMOVING NOISE FROM THE MEASURED SIGNAL AND COMPUTING THE ORIGINAL SIGNAL USING MATLAB

Firstly we measure the sound signal and then there is a traffic or the inherent noise added from the measuring equipment that is Microphone, which is computed in MATLAB

```
t=0:1:100;
x1=sin(2*pi*0.05*t);
subplot(5,1,1);
plot(t,x1);
xlabel('Time (s)');
ylabel('Amplitude');
grid on;
n1=t;
n2=fliplr(n1);
y1=rand(1,length(t));
z1=x1+y1;
subplot(5,1,2);
plot(t,z1);
title('Noisy Signal from Microphone 1');
xlabel('Time (s)');
ylabel('Amplitude');
grid on;
```

As a result signal is generated and also with a noise added with the measured sound

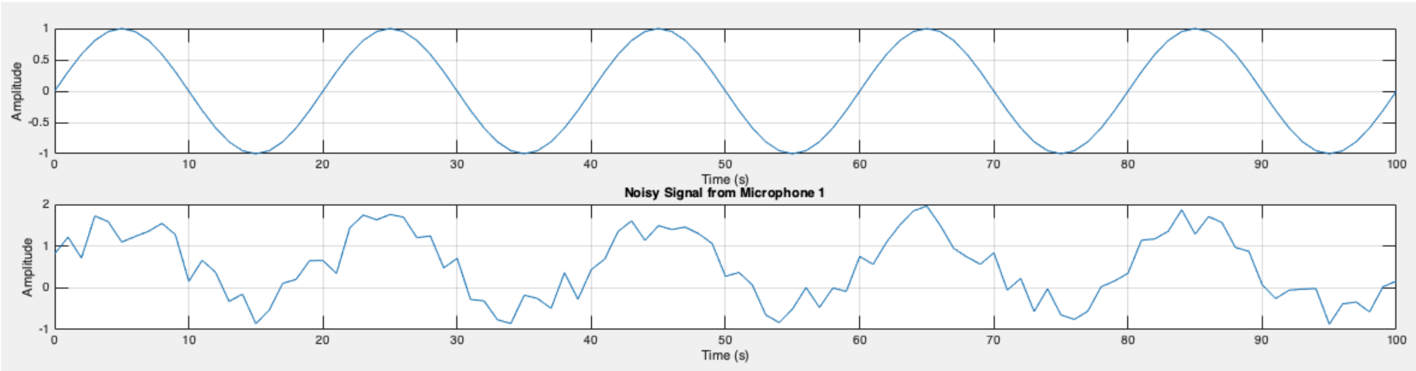

Figure 24: Output from the Microphone 1

Similarly, from the Microphone 2

```
t=0:1:100;
x2=sin(2*pi*0.05*t);
subplot(5,1,3);
plot(t,x2);
xlabel('Time (s)');
ylabel('Amplitude');
grid on;
n1=t;
n2=fliplr(n1);
y2=rand(1,length(t));
z2=x2+y2;
subplot(5,1,4);
plot(t,z2);
grid on;
title('Noisy Signal from Microphone 2');
xlabel('Time (s)');
ylabel('Amplitude');
```

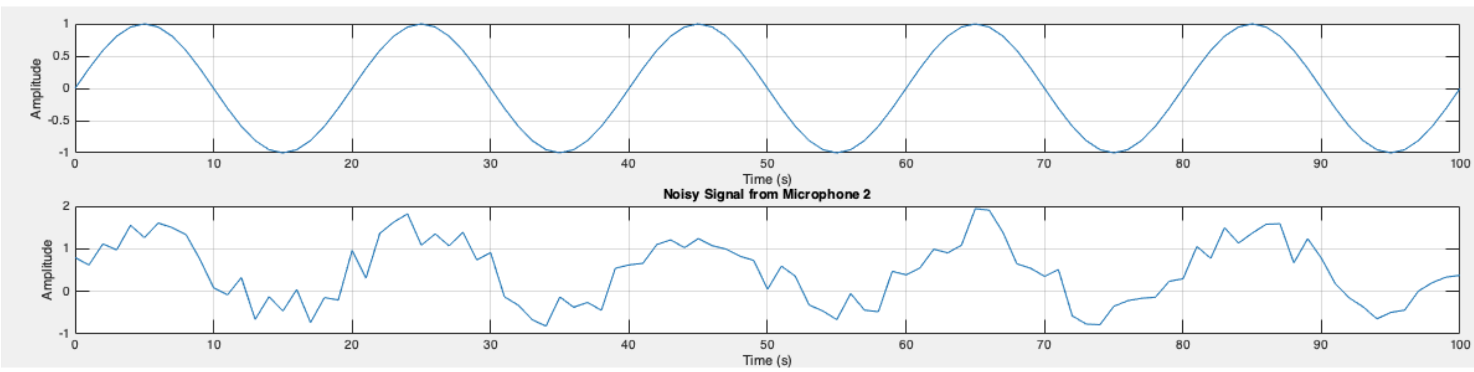

Figure 25: Output from the Microphone 2

So, after two signals have been measured from 2 microphones has been measured along with the self noise of the microphone. Hence, we use the Cross-Correlation function in order to remove the noise from the signal and we have the measured sound signal without any traffic or noise in it.

And the Output after the Cross-Correlation is shown below,

```
m=xcorr(z1,z2);
t1=t;
t2=fliplr(t);
n1=min(t1)+min(t2);
n2=max(t1)+max(t2);
n=n1:1:n2;
subplot(5,1,5);
plot(n,m);
grid on;
title('Final signal');
xlabel('Time (s)');
ylabel('Amplitude');
```

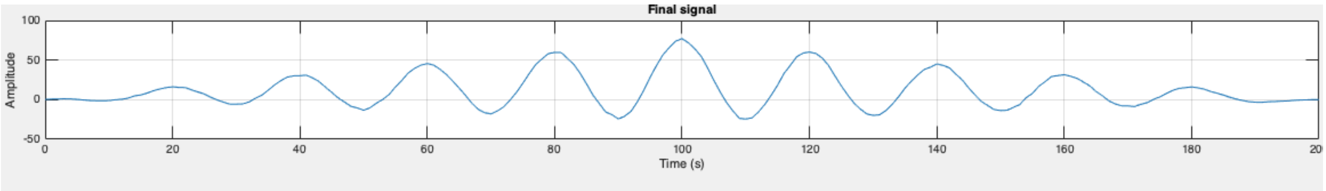

Figure 26: Measured sound signal after Cross-Correlation

9. MEASUREMENT USING THE STANDARD EQUIPMENTS

Firstly, before the the measurement is carried out, we have to know about the Sound pressure level and how to measure. As we are carrying out our experiment based on calculating the sound pressure level.

9.1 SOUND PRESSURE LEVEL

If we speak about sound level they generally apply to sound level in decibels. This may be when talking about how noisy a rock band, jet engines, or city ordinances are. What are the decibels, then? Let's take a look at sound pressure first to answer that. Sound is determined by the pressure changes in the air. The louder a sound is, the greater is the increase in air pressure. The difference here is the transition from the usual ambient pressure to the sound pressure disturbance. This pressure shift can be measured by portable devices or special-microphone computers. Sound pressure is usually expressed in pascals, which is a unit of SI (metric). A Pascal (symbol Pa) is equal to one newton force per square metre. Compared to some of the pressure units one may be familiar with, such as pounds per square inch, a pascal is "thin" For example, about 241,000 pascals, or about 241 kilopascal, are equivalent to a tire pressure of 35 pounds per square inch. The smallest intensity of sound that a human ear can detect is 20 micropascal (0.000020 Pa). It is easy to write this scientific notation as 2.0×10-5 Pa [10].

SOURCE	Sound Pressure (Pa)
Leaf rustling	0.0000632
Normal conversation	0.01
TV set at home	0.02
Passenger car as heard from roadside	0.1
Jack hammer	2.0
Jet engine as heard from 100 yards	100
Extremely loud rock band	200
Jet engine as heard from 1 yard	630

Table 2 : Sound pressure levels in Pascals(Pa)

As its evident, these numbers vary from very small numbers with four zeroes after the decimal point. A special logarithmic scale has been developed to make dealing with this variety of numbers more manageable. Which in turn points toward the Decibel Scale.

The unit decibel (symbol dB) is a logarithmic unit which expresses the ratio of two values. The decibel was named in honour of Alexander Graham Bell (1847-1922), a famous scientist. To calculate the sound pressure level (SPL) in the decibels, we use the following logarithmic formula when measuring vibration.

$$L_p = 10 \bullet \log\left(\frac{p}{p_0}\right)^2 = 20 \bullet \log\left(\frac{p}{p_0}\right),$$

Where p is the sound pressure we measure, and Po is the standard lowest sound pressure we can hear that is 2.0×10-5 Pa.

SOURCE	Sound Pressure (Pa)	Sound Pressure Level (dB)
Leaf rustling	0.0000632	10
Normal conversation	0.01	54
TV set at home	0.02	60
Passenger car as heard from roadside	0.1	74
Jack hammer	2.0	100
Jet engine as heard from 100 yards	100	134
Extremely loud rock band	200	140
Jet engine as heard from 1 yard	630	150

Table 3 : Sound pressure levels in Pascals(Pa) and Decibel (dB)

We now have, as you can see, a range of numbers that seems more manageable from 10 to 150 rather than a range from a very small number . That is an export to use convenient numbers instead of the very small to very large range.

9.2 WEIGHTING

One of the most important things you need to know about when measuring sound can be to consider the difference between noise frequency weightings. This is because picking the wrong sound level meter weighting on your sound level pressure meter could lead to your results being irrelevant for the reason and at worst. We may have noticed that certain sound meters allow you to select the weighting of frequencies you want to measure noise. The three decibel weights most widely used are 'A', 'C' and ‘Z’ [9].

9.2.1 A - FREQUENCY WEIGHTING

A weighting of the sound level meter that makes the readings correspond to a theoretical human hearing response. It is specified in different international standards such as IEC 61672 and in different national standards such as ANSI S1.4. (USA). 'A' Weighted is the most widely used frequency range that encompasses the entire 20Hz frequency spectrum up to 20 kHz. The human ear is most sensitive to sound frequencies between 500 Hz and 6 kHz (especially around 4 kHz) whereas the human ear is not very sensitive at lower and higher frequencies.

The 'A' weighting changes the sound pressure level readings to reflect the sensitivity of the human ear and is thus compulsory worldwide for potential assessments of hearing damage. Any licensed sound level meter that meets IEC 61672 is required to include at least one A-weight filter. The measurements are usually shown as dB(A) or dBA. For our practice we use A-WEIGHTING.

9.2.2 C - FREQUENCY WEIGHTING

The C-weighted frequency looks more at the influence of low frequency sounds on the human ear compared to the A-weight and is basically flat or linear between 31.5Hz and 8kHz. Measurements of peak sound intensity are rendered using weighting of the C- frequency. This is a c-weighted value for impulse noise calculation and is called C Peak. Usually the measurements are shown as dB(C) or dBC.

9.2.3 Z - FREQUENCY WEIGHTING

Z-weighted is the flat frequency response of 8Hz to 20kHz (+ /- 1.5dB), that's the real noise produced for the human ear without any weighting (Z for zero). Also used in the study of octave bands and to evaluate ambient noise. Measures taken are shown as dB(Z) or dBZ

The standard IEC 61672 sound level meter defines the output and tolerances to be used for the frequency weighting curves.

Frequency (Hz)	**63**	**125**	**250**	**500**	**1k**	**2k**	**4k**	**8k**	**16k**
A-weighting (dB)	– 26.2	– 16.1	– 8.6	– 3.2	0	1.2	1	-1.1	– 6.6
C-weighting (dB)	– 0.8	– 0.2	0	0	0	– 0.2	– 0.8	– 3.0	– 8.5
Z-weighting (dB)	0	0	0	0	0	0	0	0	0

Table 4: Weighting Frequency ranges

10 . EXPERIMENTAL SETUP

There is a methodology used to set up the equipments in order to achieve the desired results. So we need the setup inside an anechoic chamber and a signal analyser and a computer On B&K Pulse Software to compute or to read the acquired results as shown in the Figure 27.

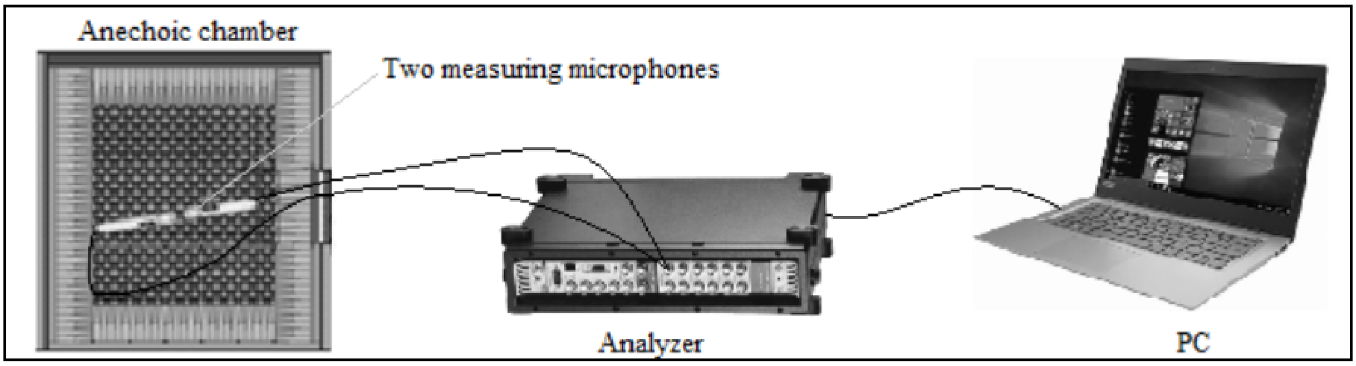

Figure 27: Experimental setup

The microphones are positioned so that they detect the same signal of the acoustic field also we use the two same microphones to get the results as similar as the other one. This can be achieved by positioning the microphones face-to-face, such as the setup used for measuring sound strength. The microphone diaphragm should be close together compared to the maximum frequency of interest wavelength: a good guide is less than a quarter of the wavelength. Thus, a gap of less than 5 mm would match typical applications of sound level meters. The microphones can also be for lower frequencies set in a side-by-side setup as shown in Figure 28. The analyser settings are set accordingly in Pulse software as shown in the Figure 29.

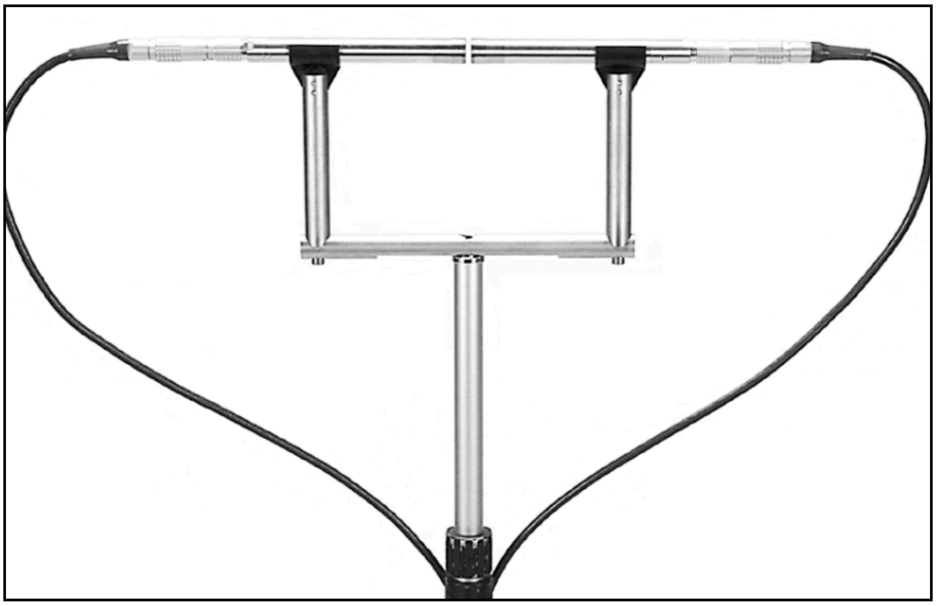

Figure 28: Positioning of Microphone

The low-noise 20 dB gain preamplifier is usually used. This assembly has a supplier-specified range of 6.5 dB-110 dB dynamic measurement. This means the microphone can not be used to test less than 10 dB consistently. ,If the noise in the frequency range of the given measurement is very low, then only the noise of the microphone itself will be measured, or does this particular microphone require sound pressure levels to be measured below 5 dB. There is no appropriate microphone for those serious sound pressure low levels below 5 dB measurements. Special measuring methods should be used in this situation. So after the set up there are is an addition of additive noise called self noise or the inherent noise of the microphone itself to the measured sound signal, which in turn affects the accuracy, there are different ways to get the additive noise out of the signal. The Cross-Correlation method is used in this practise[12].

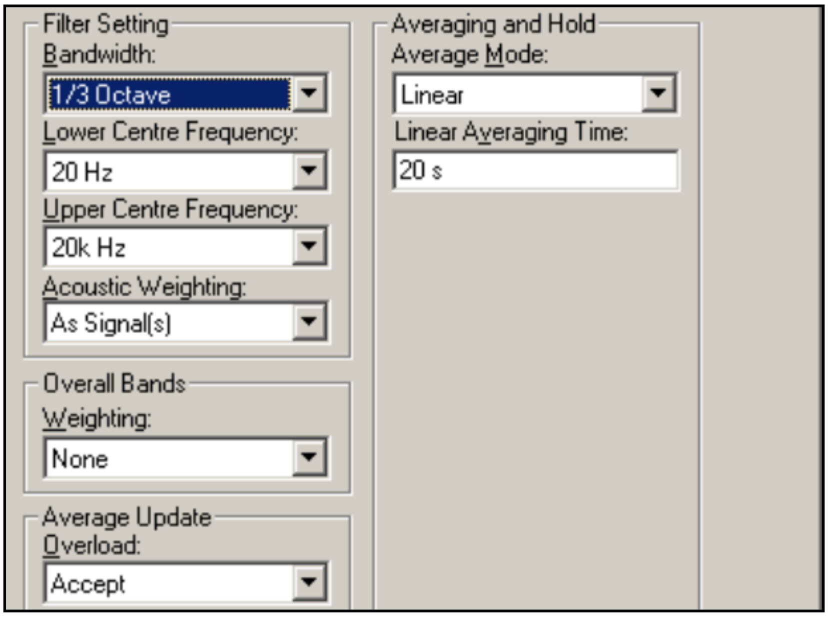

Figure 29: Analyser settings

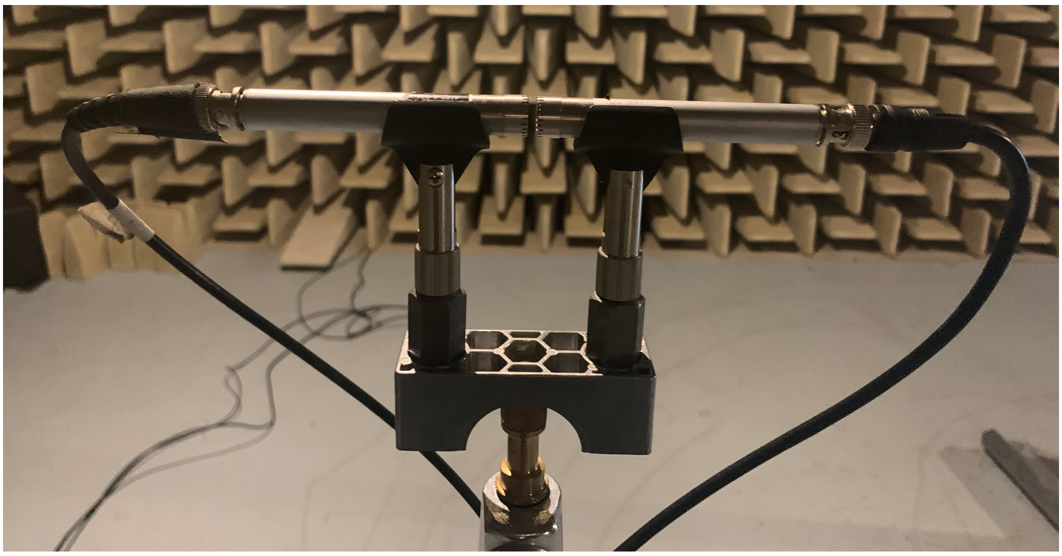

Figure 30: Positioning of Microphone in Anechoic chamber

We perform this measurement in two different fields, firstly we set up the microphone as shown in the Figure below inside an anechoic chamber freely with open doors and there will be an influence of ambient noise and also other traffic noise being measured. The next set up will be creating Anechoic chamber inside an Anechoic chamber with the help of boxes and the sound absorbing sponges and then the microphone is kept inside those box and the box is closed with a lining of sponge, ignored to make sure that there is no influence of the ambient noise or any other noise which may affect the measurement as well as the accuracy and the set up is as shown in the Figure below. The Measurements are performed twice in 2 different setup in order to make sure that the desired results can be achieved even with the standard measurement equipments. This is the setup of the experiment performed and then results are computed on computer in Pulse software in the plots which are discussed below[10].

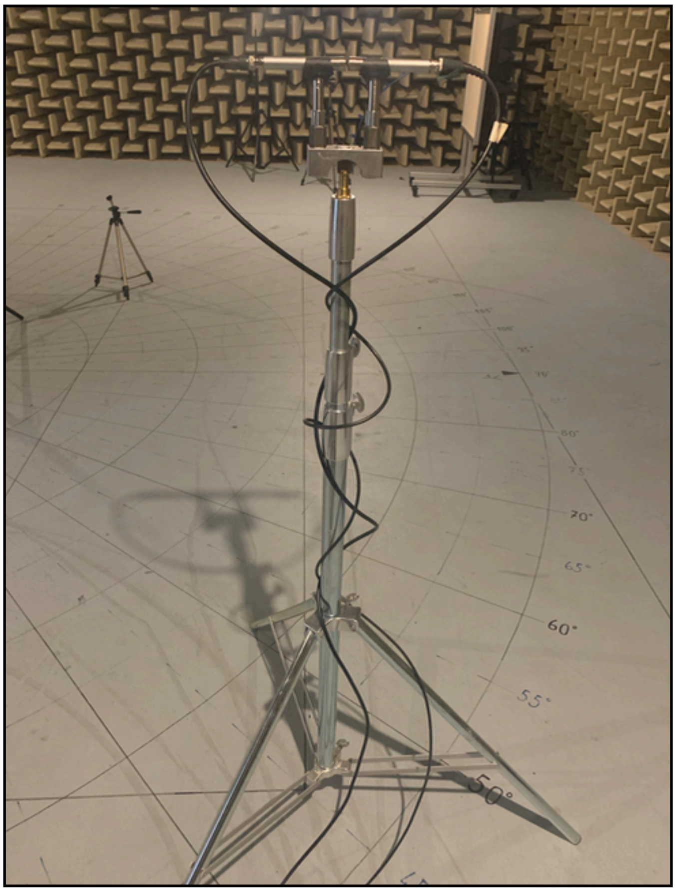

Figure 31: Setup freely inside anechoic chamber

Figure 32:Anechoic chamber inside an Anechoic chamber

11. ANALYSIS OF THE RESUTS

As discussed in the experimental setup, we performed the measurement in two different ambience first one free in Chamber and then creating an anechoic chamber inside an anechoic chamber with the help of wooden boxes and sound absorbing sponge as a lining on the walls of the wooden box. The results are obtained in the form of graphs and are as follows,

11.1 FREE IN CHAMBER

The results are plotted for the sound levels in decibels (dB) from Microphone 1 , Microphone 2 and the Cross-Correlation of the signals from those two microphones with reference to Frequency (Hz). The Graph specifies Frequency in Hertz (Hz) on X- axis and Decibel (dB) on Y-axis. The sound samples were measured for 20 seconds.

11.1.1 SOUND SIGNAL FROM MICROPHONE 1

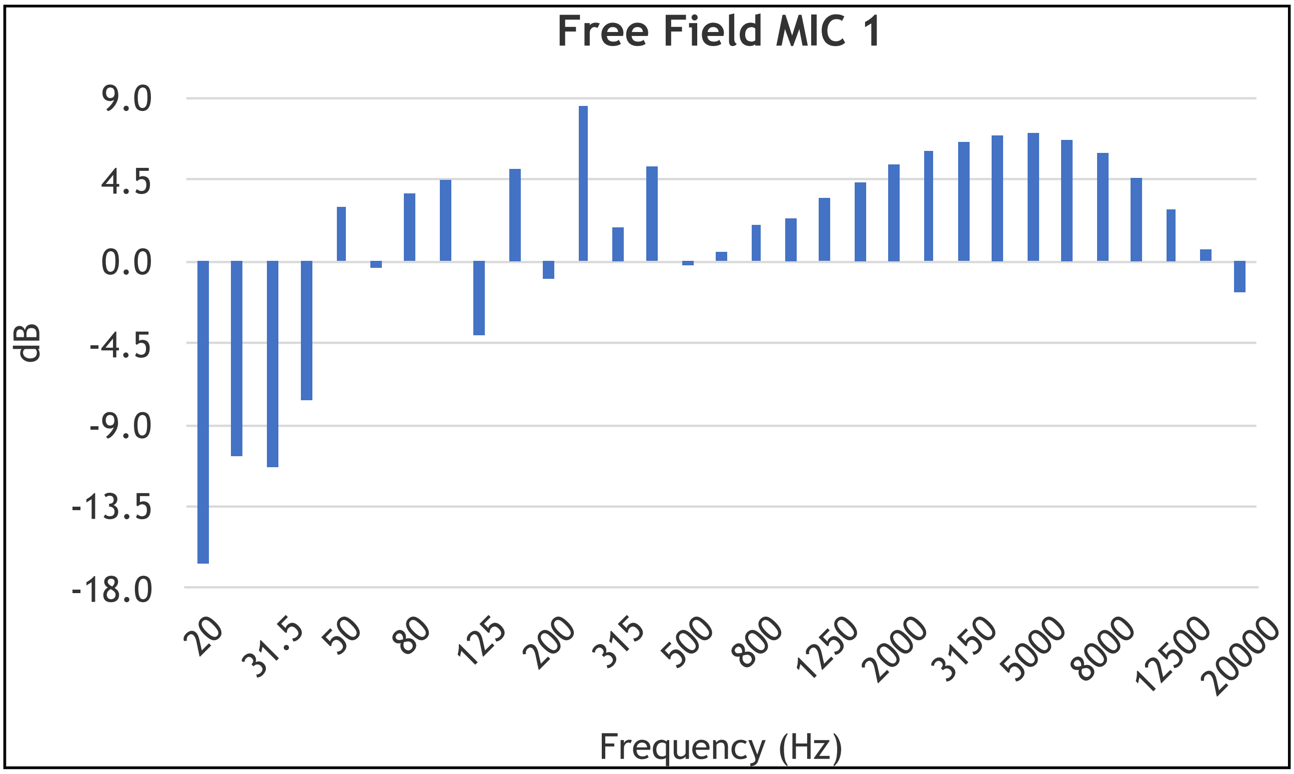

Figure 33: Sound signal from Microphone 1 (free)

11.1.2 SOUND SIGNAL FROM MICROPHONE 2

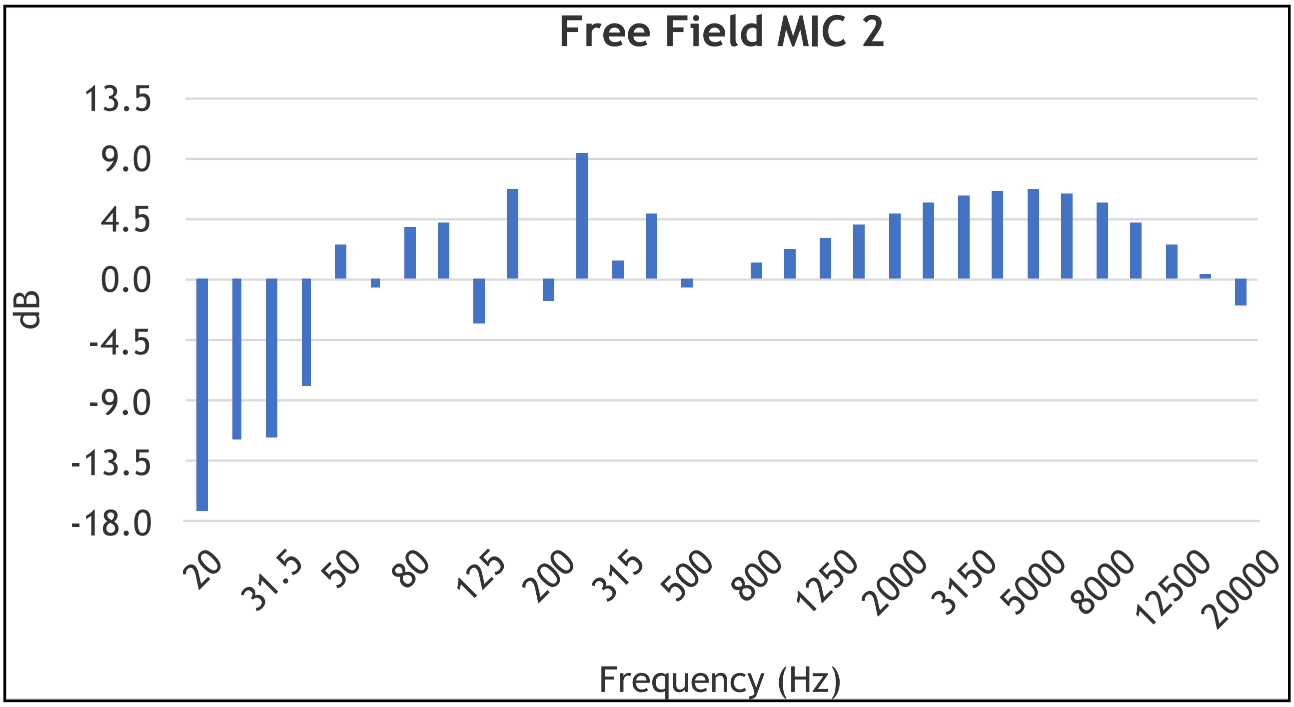

Figure 34: Sound signal from Microphone 2 (free)

11.1.3 CORRELATION SIGNAL OF MICROPHONE 1 & 2

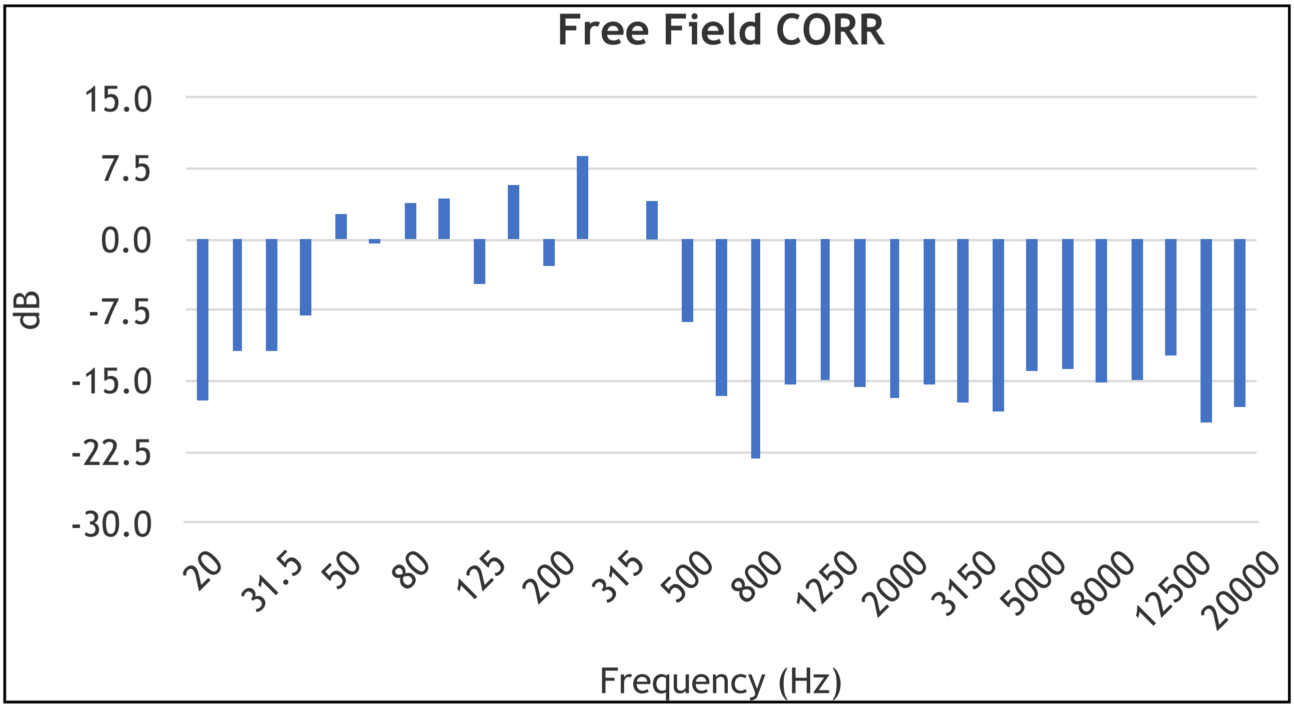

Figure 35: Correlation of Microphone 1 & 2(free)

11.1.4 PLOT MENTIONING SIGNALS OF MICROPHONE 1, 2 & CORRELATION

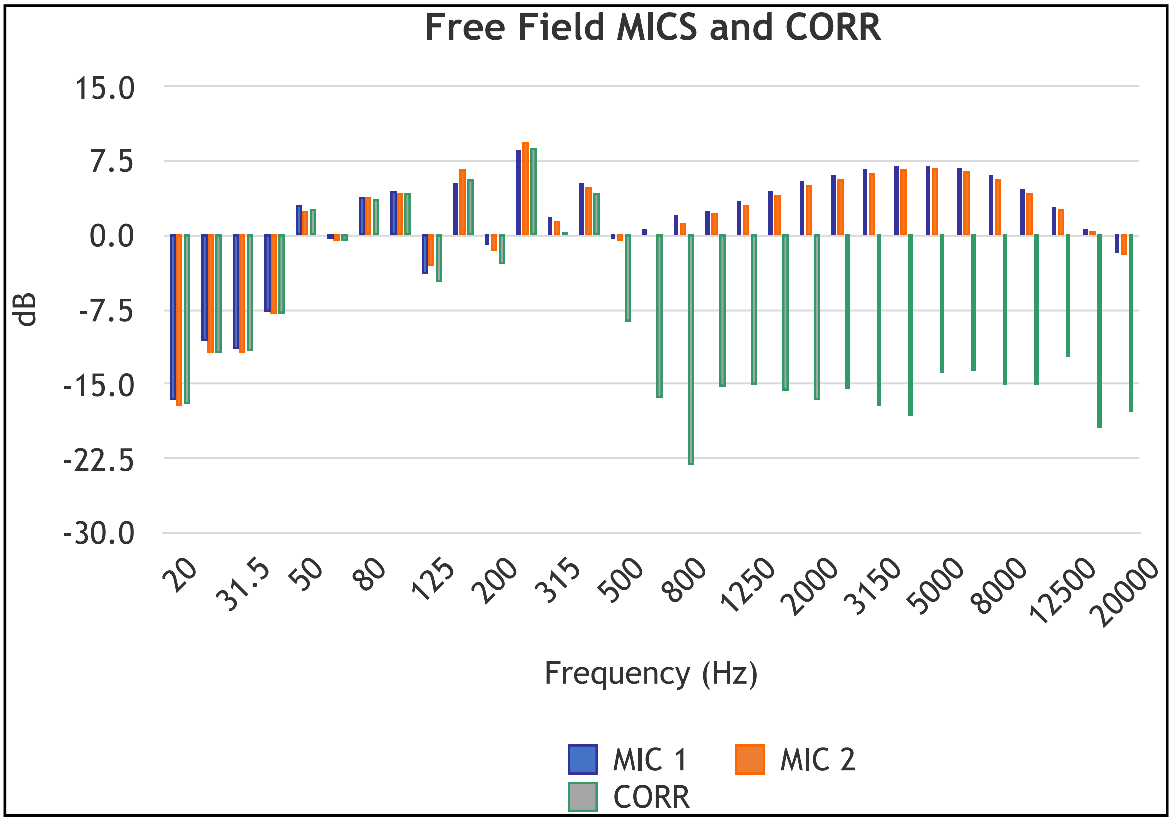

Figure 36: Plot mentioning sound signals from Mic 1, 2 and Corr (free)

As the plots above are the results from the measurements performed freely inside an anechoic chamber, and the average of sound measured from Microphone1 is 18.6 dB, and from Microphone2 is 18.5 dB and then by correlation of these two obtained signals the average sound level measured is 13.8 dB.

11.2 CLOSED INSIDE THE BOX INSIDE CHAMBER

Similarly, the results are plotted for the sound levels in decibels (dB) from Microphone 1 , Microphone 2 and the Cross-Correlation of the signals from those two microphones with reference to Frequency (Hz). The Graph specifies Frequency in Hertz (Hz) on X- axis and Decibel (dB) on Y-axis. The sound samples were measured for 20 seconds.

11.2.1 SOUND SIGNAL FROM MICROPHONE 1

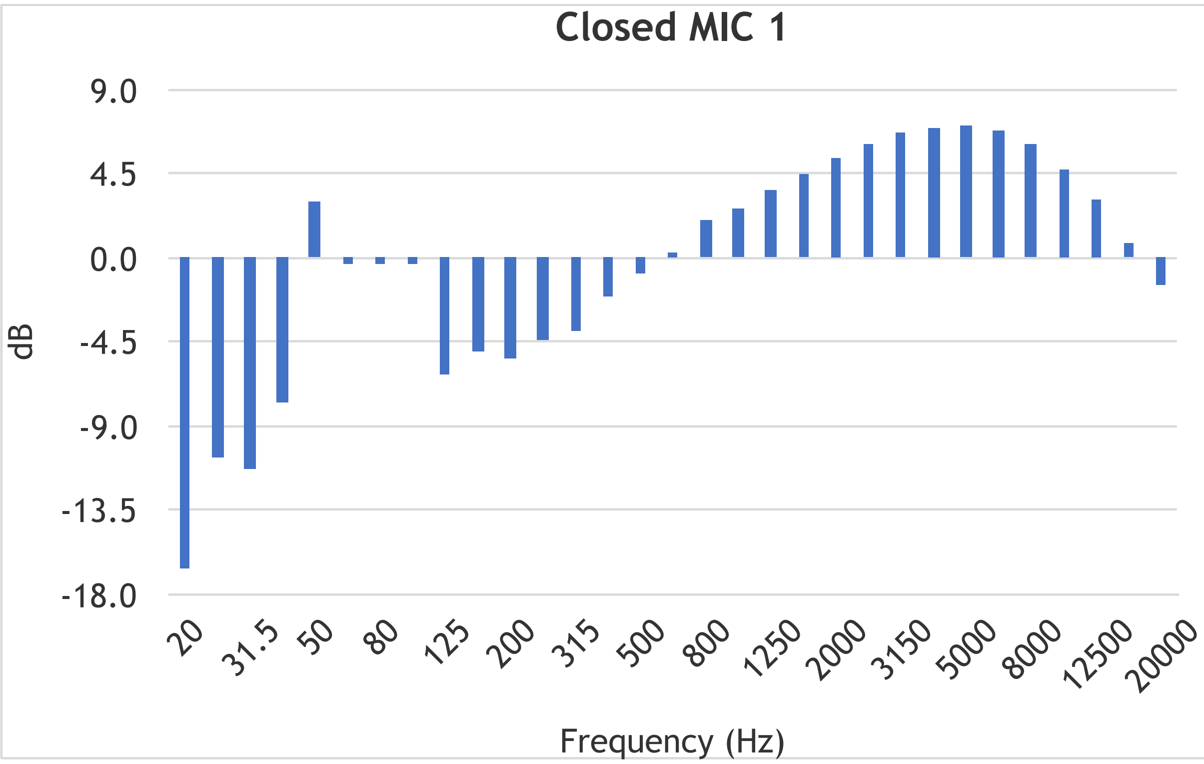

Figure 37: Sound signal from Microphone 1(closed)

11.2.2 SOUND SIGNAL FROM MICROPHONE 2

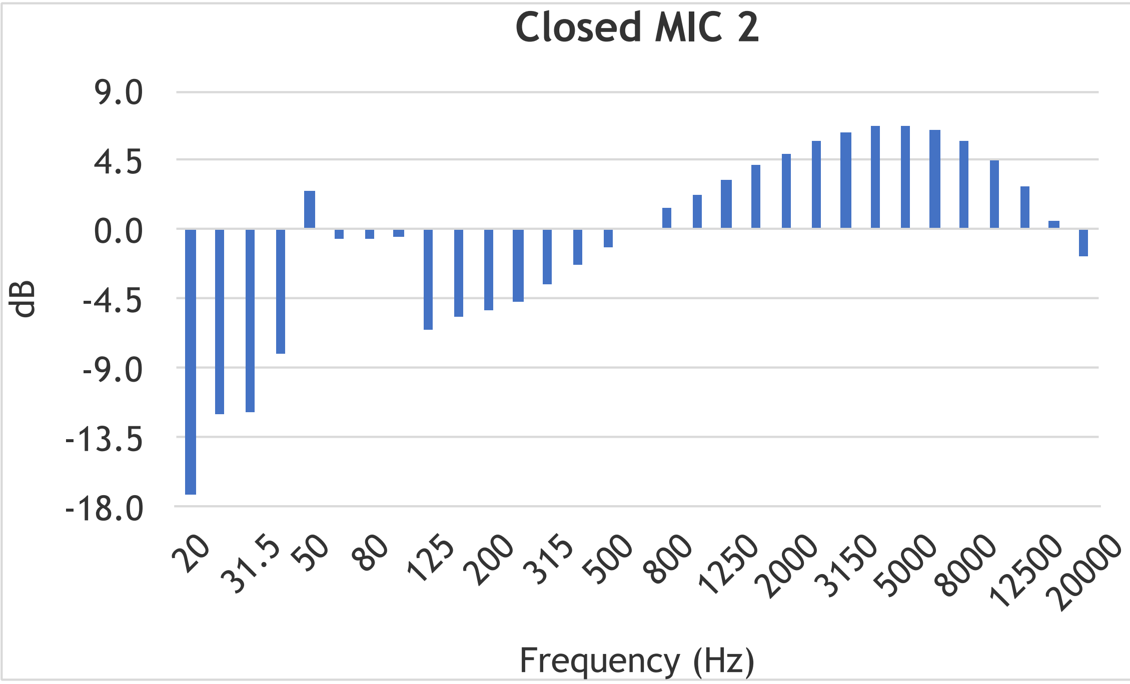

Figure 38: Sound signal from Microphone 2(closed)

11.2.3 CORRELATION SIGNAL OF MICROPHONE 1 & 2

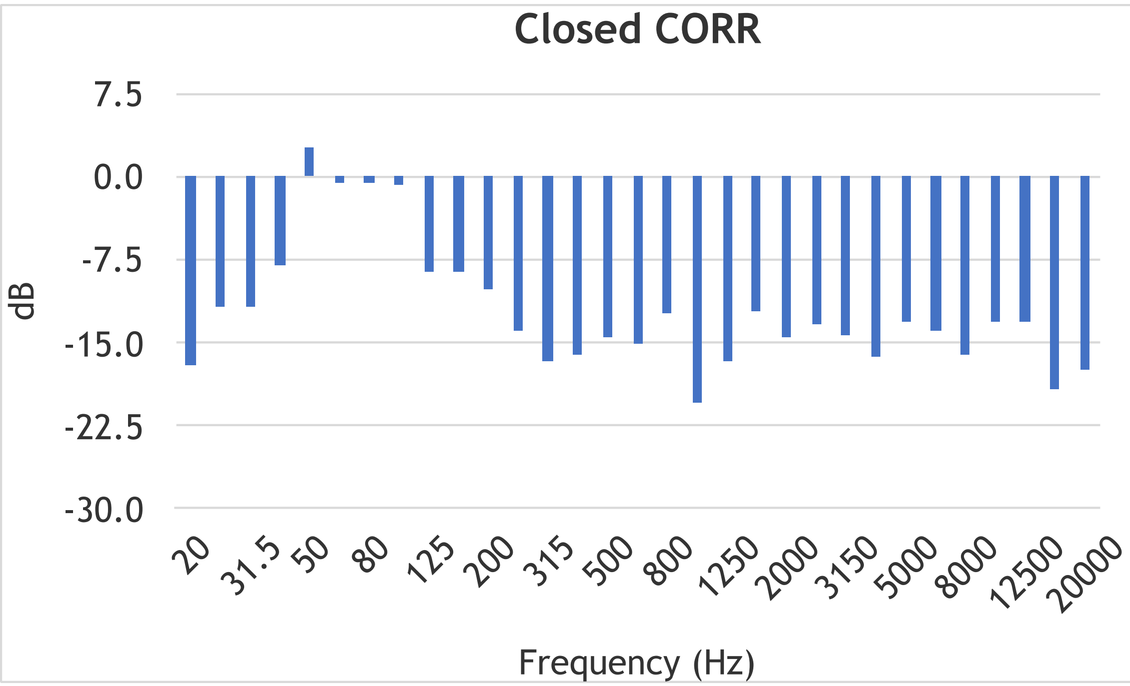

Figure 39: Correlation of Microphone 1 & 2(closed)

11.2.4 PLOT MENTIONING SIGNALS OF MICROPHONE 1, 2 & CORRELATION

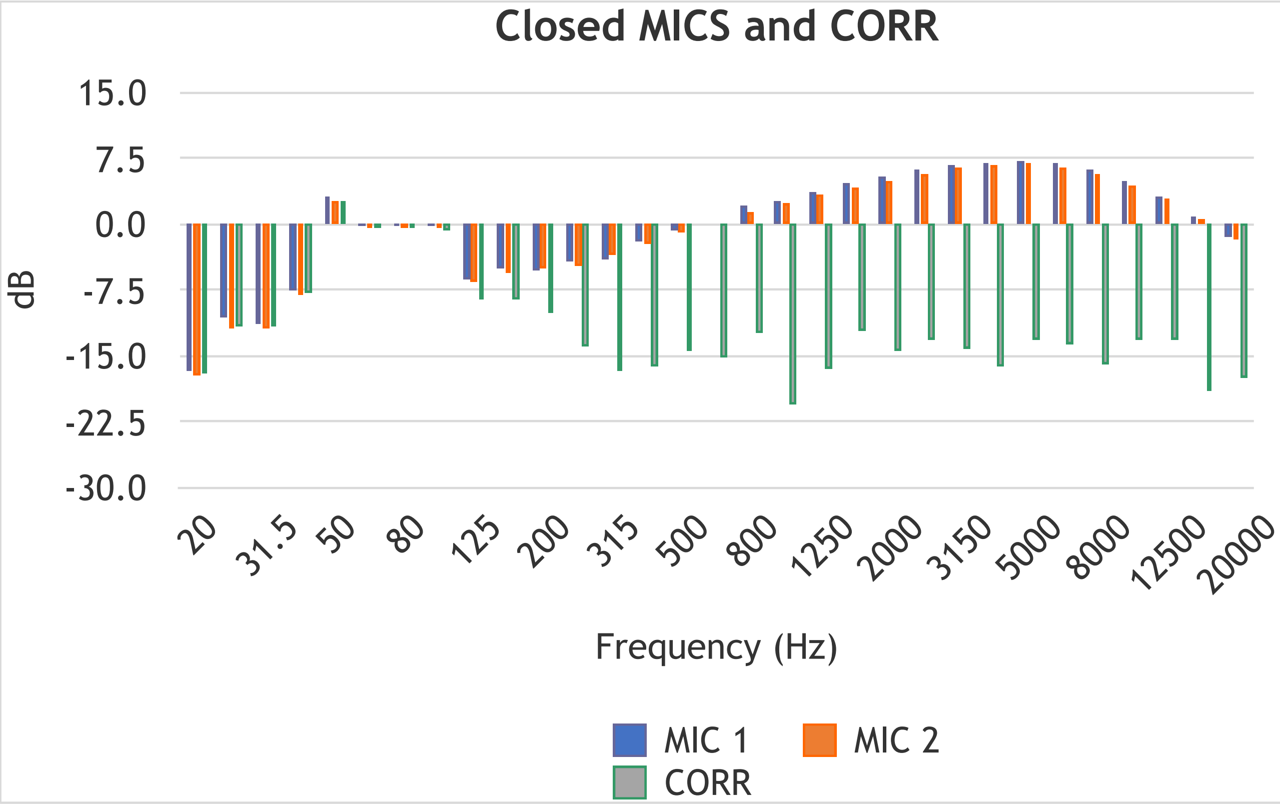

Figure 40: Plot mentioning sound signals from Mic 1, 2 and Corr (closed)

As the plots above are the results obtained from the measurements performed in a closed wooden box inside an anechoic chamber, and the average of sound measured from Microphone1 is 17.5 dB, and from Microphone2 is 17.2 dB and then by correlation of these two obtained signals the average sound level measured is 7.7 dB.

CONCLUSION

Noise is always a Barrier in sound measurement where some of the noise can be avoided with simple precautions but there are some noise which are unavoidable which in turn affects the accuracy of the sound measurement. When speaking about the sound measurement below the below the minimum measurable sound pressure level, There comes the picture of self noise/inherent noise of the microphone will be added to the measured sound signal, In general practice to measure low level of sound pressure, its quite obvious to choose advanced measurement equipments like a specialised low noise measurement microphone in order to achieve the results and the obtained results to be so accurate but to prove the low level sound measurement is even possible with standard measurement equipment and a proper setup is achievable. The measurement setup was built and measured in two different fields to make sure the comparison brings out the accuracy. Firstly the sound measurement was done in freely inside an anechoic chamber and the results were recorded, and similarly an anechoic chamber inside an anechoic chamber was built with the help of wooden boxes with an inner layer of sound absorbing sponge and the microphones were placed inside it and then measurement was done and results were obtained. Also to have precise output from the microphones , a similar kind of microphones were used and placed facing each other with a small distance to measure the sound. The measured sound with added intent noise was removed with a technique of Cross-correlation, to eliminate the additive noise and achieve accuracy in measurement.

This Thesis focused on an improved method for measuring the very low levels of sound pressure using standard measurement equipment . This approach was based on self noise/Inherent noise elimination using the Cross-correlation function of two signals measured by two measuring microphones. The application of this method led to a significant reduction of the level of background noise and self noise of the measurement equipment . For this purpose, the Cross-correlation effect approach leads to more precise measurements of very low sound pressure level. Hence it can be concluded that Low level sound measurement even with a standard measurement equipment is achievable.

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